SC Tehsino International SRL
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VoIP Phone

TVP 2208P

User Manual

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Safety Notices

Please read the following safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

- Please use the external power supply that is included in the package. Other power supplies may cause damage to the phone, affect the behavior or induce noise.
- Before using the external power supply in the package, please check with home power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, do not use it, it may cause fire or electric shock.
- The plug-socket combination must be accessible at all times because it serves as the main disconnecting device.
- Do not drop, knock or shake it. Rough handling can break internal circuit boards.
- Do not install the device in places where there is direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposure the phone to high temperature, below 0℃ or high humidity. Avoid wetting the unit with any liquid.
- Do not attempt to open it. Non-expert handling of the device could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock and breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with a soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug or phone line, it may cause an electric shock.
- Do not install this phone in an ill-ventilated place.
- You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

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Table of Content

1. INTRODUCING TVP-2208P VOIP PHONE 5

1.1. THANK YOU FOR YOUR PURCHASING TVP-2208P 5
1.2. DELIVERY CONTENT 5
1.3. KEYPAD 6
1.4. PORT FOR CONNECTING 7

2.INITIAL CONNECTING AND SETTING 8

2.1. CONNECT THE PHONE 8
2.2. INITIAL SETTING 9
2.2.1. PPPoE mode 9
2.2.2. Static IP mode 10
2.2.3. DHCP mode 10

3. BASIC FUNCTIONS 11

3.1. BASIC OPERATION 11
3.1.1. Accepting a call 11
3.1.2. Making a call 11
3.1.3. Ending a call 12
3.1.4. Transferring a call 12
3.1.5. Calling Hold 13
3.1.6. 3-way conference call 13
3.1.7. Switchboard Operator feature 13
3.1.8. Call records 13
3.2. THE HIGH-LEVEL OPERATION 14
3.2.1. SMS function 14
3.2.2. Memo function 14
3.2.3. SpeedDial function 14
3.2.4. Realize Secondary Dial by Dialing for only one time 14
3.2.5. Phonebook prefix function 15
3.2.6. Function key 15
3.2.8. join call 16
3.2.9. redial/unredial 16
3.2.10. click to dial 17

4. SETTING 17

4.1. INTRODUCTION OF CONFIGURATION 17
4.1.1. Ways to configure 17
4.1.2. Password Configuration 17
4.2. SETTING VIA WEB BROWSER 17
4.3. CONFIGURATION VIA WEB 18
4.3.1. BASIC 18
4.3.1.1. Status 18
4.3.1.2. Wizard 19
4.3.1.3. Call Log 20
4.3.1.4. MMI SET 21
4.3.2. Network 21
4.3.2.1. WAN Config 21
4.3.2.2. LAN Config 23
4.3.2.3. Qos Config 24
4.3.2.4. Service Port 26
4.3.2.5. DHCP SERVER 26
4.3.2.6. SNTP 27
4.3.3. VOIP 29
4.3.3.1. SIP Config 29
4.3.3.2. IAX2 Config 31

SC Tehsino International SRL
Add: Str. Gladitei Nr.42, Bl. T6, Ap. 1201, Sector 4, Bucharest
Tel: 0040212114147, Fax: 004021242 4136
www.tehsino.ro

4.3.3.3. STUN Config 32
4.3.3.4. DIAL PEER setting 34
4.3.4. Phone *37*
4.3.4.1. DSP Config 37
4.3.4.2. Call Service 38
4.3.4.3. Digital Map Configuration 39
4.3.4.4. Phone Book 40
4.3.5. Maintenance *41*
4.3.5.1. Auto Provision 41
4.3.5.2. Syslog Config 42
4.3.5.3. Config Setting 43
4.3.5.4. Update 44
4.3.5.5. Account Config 44
4.3.5.6. Reboot 45
4.3.6. Security *46*
4.3.6.1. MMI Filter 46
4.3.6.2. Firewall 46
4.3.6.3. NAT Config 48
4.3.6.4. VPN Config 50
4.3.7. Logout *52*
4.4. SETTINGS VIA PHONE'S KEYBOARD. 53
4.4.1. How to set via the phone's keyboard. 53
4.4.2. Phone menu 53

5. APPENDIX 54

5.1. SPECIFICATION 54
5.1.1. Device specification 54
5.1.2. Voice Features 54
5.1.3. Network Features 54
5.1.4. Maintenance and Management 55
5.2. DIGIT-CHARACTER MAP TABLE 55

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1. Introducing TVP-2208P VoIP Phone

1.1. Thank you for your purchasing TVP-2208P

Thank you for your purchasing TVP-2208P, TVP-2208P is a full-feature telephone that provides voice communication over the same data network that your computer uses. This phone functions are not only like a traditional phone, allowing to place and receive calls, and enjoy other features that traditional phone has, but also it owns many data services features which you could not expect from a traditional telephone.
This guide will help you easily use the various features and services available on your phone.

1.2. Delivery Content

Please check whether the delivery contains the following parts:

The base unit with display and keypad
The handset
The handset cable
The power supply
The Ethernet cable

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1.3. Keypad

The numeric keypad with the keys 0 to 9, *, and # is used to enter

Digits and letters, additionally, the following keys are available:

Key mapping:

Key	Key name	Function Description
	Menu	In idle state，press the MENU key to call up the menu.
	Phone Book	In idle mode, press the Phone Book key to check the record list and add new records and revise the record. Press this key again will return to idle mode.
	Callers	In idle/pickup/calling mode, press the Callers key to Check the Income/Outgoing/Missed calls records. Press this key again will return to idle mode
	LED	LED blinks to remind user new voicemail.
	Soft1 Soft2 Soft3	Use the Softkey to realize the various functions, like DND/SDial/Memo/Enter/Next/Del/Save/Quit/Dial/Edit/Redial/EDial, etc.
	Navigation Key	In idle mode, press DOWN key to check the phone setting parameters. Such as network mode, local IP and local Gateway IP address. Press UP key to check the missed call log, press LEFT key to check the incoming call log, press the RIGHT key to browse the lines' register status. When you pick up the handset or during calling, you can use this key to turn up or turn down the handset volume; when a call comes, you can use this key to adjust ring volume; you also can use this key to choose item in the menu, callers or phone book.
	MWI	Use this key to read old or new message. User can replace it with other functions, like pickup key, record key, etc.
	SMS	Use this key to send and check the message; Support the multilanguage input.

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	Transfer	Use the key to realize blind transfer or attended transfer please refer to **3.1.4.**-call transfer for more details).
	Release	In menu, use this key to return to the idle state. And when you are talking, you can use it to end the call and then return to the idle state.
	Hold	Temporarily hold the active call during the talking; press the key again to unhold the call. **(3.1.5-Calling Hold).** In the idle state, press this key to enable the DND, press the key again to disable the DND.
	mute	Press this key in calling mode, you can hear the other side, and the other side can not hear you
	R/send	In the hook off /hands-free mode, use the key to dial the last call number; use this key to make a quick dial as soon as you select your desired number in phone book or callers. In the idle state, use the key to browse the outgoing call log.
	Handfree	Enter into hands-free mode.

1.4. Port for connecting

POWER	Power switch	Select ON/OFF
DC	Power port	Output: 5V/1.0A
LAN	Network port	Connect it to PC
WAN	Network port	Connect it to Network

The phone has two Network ports: The WAN port and the LAN port. Before you connect the power source, please carefully read Safety Notices of this user manual.

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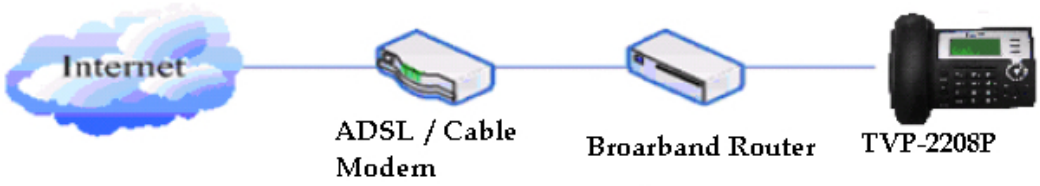
2.Initial connecting and Setting

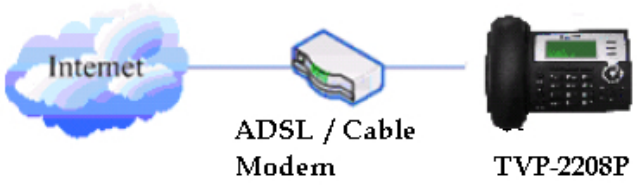
2.1. connect the phone

Step 1: Connect the IP Phone to the corporate IP telephony network. Before you connect the phone to the network, please check if your network can work normally.

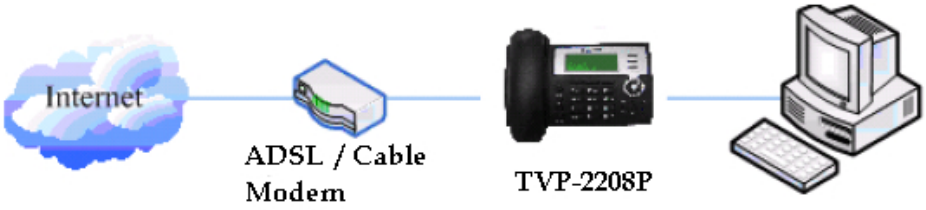
You can do this in one of two ways, depending on how your workspace is set up.

Direct network connection—by this method, you need at least one available Ethernet port in your workspace. Use the Ethernet cable in the package to connect WAN port on the back of your phone to the Ethernet port in your workspace. Since this VoIP Phone has router functionality, whether you have a broadband router or not, you can make direct network connect. The following two figures are for your reference.





Shared network connection—Use this method if you have a single Ethernet port in your workspace with your desktop computer already connected to it. First, disconnect the Ethernet cable from the computer and attach it to the WAN port on the back of your phone. Next, use the Ethernet cable in the package to connect LAN port on the back of your phone to your desktop computer. Your IP Phone now shares a network connection with your computer. The following figure is for your reference.



Step 2: Connect the handset to the handset port by the handset cable in the package.

Step 3: connect the power supply plug to the DC port on the back of the phone. Use the power cable to connect the power supply to a standard power outlet in your workspace.

Step 4: push the on/off switch on the back of the phone to the on side, then the phone's LCD screen displays "Initializing…Wait LogOn…". Later, a ready screen typically displays the greeting word and the date, time and softkey name.

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If your LCD screen displays different information from the above, you need refer to the next section "Initial setting" to set your network online mode.

If your VoIP phone registers into corporate IP telephony Server, your phone is ready to use.

2.2. Initial Setting

This VoIP Phone provides you with rich function and parameters setting. If you have enough knowledge about network and SIP protocol, it is better for you to understand many parameters. But if you know little about network and SIP protocol, you can also easily make initial setting according to the following steps to enjoy rapidly high quality voice and low cost from this VoIP Phone.

Before make initial setting, please check if your corporate IP telephony network can work normally, and you have finished "connect the phone".

This VoIP Phone Supports DHCP by default. It will receive an IP address and other network-related settings (Netmask, IP gateway, DNS server) from the DHCP server. If your network supports DHCP, you can connect this VoIP Phone directly to the network. If your network doesn't support DHCP, you need change this VoIP Phone's network connection setting. According to the following steps, change this VoIP Phone's DHCP network connection setting into PPPoE or static IP which your network supports at present.

2.2.1. PPPoE mode

1. Please prepare your PPPoE account name and password
2. Press the MENU key, and press the key for twice, the LCD screen will display "**3 Network**". Then press the Soft2(Enter) key, the LCD screen will display "**1 WAN**".
3. Press the Soft2(Enter) key, and press the key for twice, the LCD screen will display "**3 PPPoE Set**".
4. Press the Soft2(Enter) key, the LCD screen will display "**1 Account**", then press the Soft2(Enter) key, there will display the account information; Press the Soft2(Edit) key and press the Soft1(Del) key to delete and input your PPPOE account number then press the Soft2(Save) key to confirm. With "**Saved**" displayed, screen will jump to show the account information currently.
5. Press the Soft3(Quit) key to return to the previous menu, then press the key, the LCD screen will display "**2 Password**". Then press the Soft2(Enter) key, there will display the password information (it will replaced by *). Press the Soft2(Edit) key and press the Soft1(Del) key to delete and input your PPPoE's password and confirm it by the Soft2(Save) key, With "**Saved**" displayed, screen will jump to show the password information currently(it will replaced by *).
6. Press the Soft3(Quit) key for twice and press the key, the LCD screen will display "**1 Net Mode**". Press the Soft2(Enter) key and then press the soft2(Edit) key, you can choose the network mode what you want by the navigation key. Now please choose the PPPoE mode by the key, the LCD screen will display "**<>PPPoE**", With Soft2(Save) pressed again，screen will show "**Saved**" and then jump to show the net mode currently.
7. pressing Soft3 (Quit) four times/MENU key/Release key to quit to stand-by status and pressing to shows "**PPPoE**"，phone tries to connect server to get IP. If there is shown "**Negotiating…**", it shows that the phone is trying to access the PPPoE Server, else it shows that the phone has already get IP with PPPoE.

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2.2.2. Static IP mode

1. Prepare the network's parameters first. IP Address，Netmask， Default Gateway and DNS server IP address are needed. Please contact the service provider or technician of network.
2. Pressing MENU key，and key for twice，screen shows"**3 Network**", then pressing Soft2(Enter)，screen will show "**1 WAN**".
3. Pressing Soft2 (Enter)，then pressing key, screen shows"**2 Static Set**".
4. Pressing Soft2(Enter) to make screen show"**1 IP**"， press Soft2(Enter) and then press Soft2(Edit) again, and Soft1(Del) to delete old parameter. Input your IP address and press Soft2 (Save). After "**Saved**" shown, screen will jump to show the IP information currently.
5. Press key to show "**2 Netmask**". Press Soft2(Enter) and press Soft2(Edit) again，and then use Soft1(Del) to delete. Input your Netmask and press Soft2 (Save). After "Saved" shown, screen will jump to show the Netmask information currently.
6. Press key to show "**3 Gateway**". Press Soft2(Enter) and press Soft2(Edit) again，and then use Soft1(Del)to delete，Input your gateway and press Soft2(Save). After "Saved" shown, screen will jump to show the gateway information currentlly.
7. Press key to show"**4 DNS**". Press Soft2(Enter) and press Soft2(Edit) again，and use Soft1(Del) to delete. Input your DNS server address and press Soft2 (Save). After "Saved" shown, screen will jump to show DNS information.
8. Press twice Soft3 (Quit) quitting. With key pressed，screen shows"**1 Net Mode**". Press Soft2(Enter) and press Soft2(Edit) again, and key，screen shows"**<>Static**"; with Soft2(Save) pressed，screen shows "**Saved**" and then shows the net mode currently.
9. pressing Soft3 (Quit) four times/MENU key/Release key to quit to stand-by status and pressing to show "**Static**". If screen shows the IP address and gateway which are set just now, it shows that Static IP mode is taken effect.

2.2.3. DHCP mode

1. Press the MENU key, and press the key for twice, the LCD screen will display "**3 Network**". Then press the Soft2(Enter) key, the LCD screen will display "**1 WAN**".
2. Press Soft2 (Enter) to show "**1 Net Mode**". After pressing Soft(Enter) and Soft2(Edit)，using navigation
   key to select until screen shows "**<>DHCP**". Press Soft2(Save)，With "saved" displayed, screen will jump to show the net mode currently.
3. Press Soft3 (Quit) four times quitting to stand-by status. Press key to show "**DHCP**"，if there is "**Negotiating…**"shown on screen, it shows that phone is keep trying to search DHCP server or get IP; If there is IP address displayed, it shows that DHCP mode has been taken effect.

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3. Basic Functions

3.1. Basic operation

3.1.1. Accepting a call

TVP-2208P will ring to indicate you when there is call incoming，below is ways to answer call：

- Pick up handset to accept incoming calls.
- Press the button
- If you need switch from a hands-free call to handset, please pick up the handset directly.
- If you need switch from a handset call to hands-free, please press the button, and then hang up the handset.

3.1.2. Making a call

- Quick-dialing

In idle mode, input the called number, and press Soft3(Dial) key or

button, phone will dial the call and use hands-free automatically.

- Use handset

Hook off (screen will show the current using line, or you could use key to select), after getting dialing tone, you could begin to dial number. After finishing it, press # and TVP-2208P will send the number and call the number. When you hear a ringback tone and screen shows the callee's number, it shows that the person you called is ringing. If callee answers the call, you can begin to talk and your phone will keep showing callee's number and counting time. Just hang up to finish talk.

- Use hands-free

Press （screen will show the currnet using line, or you could use key to select）, after getting dialing tone, you could begin to dial number. After finishing it, press # and TVP-2208P will send the number and call the number. When you hear a ringback tone and screen shows the callee's number, it shows that the person you called is ringing. If callee answers the call, an begin to talk and your phone will keep showing callee's number and counting time. Press again to finish talk.

- Use the phone book

press

in stand-by mode, and then press Soft2(Enter)，you will access to phonebook. If there are many persons records stored in the directory, you can use to search the person which you want to contact. Press to forward，and press to backward. Press Soft2 (Dial) or press

to dial the current number shown on the screen.

- Use Callers

Press the

key in stand-by mode, then select your desired phone number in callers by the key, Press to forward，and press to backward. Press Soft2 (Dial) or press

to dial the current number shown on the screen.

- Use the R/send key

Please pick up or press the key. After you hear dialing tone, please press the

key or press the Soft3(Redial) to dial the last phone number. Note: after you reboot the phone, the phone will delete callers and Redial will be invalid.

- Multi-line calls TVP-2208P

supports 2 SIP lines max，that is user could use 2 SIP accounts to register and make calls. System will use SIP 1 as default line to call. User can use the to select the line to call.

There are most two calls at the same time. Screen will display the incoming call number when user is keep talking. You can press the Soft1 (Answer) to accept it, and hold the first one (if you want to use this function, you need enable Call Waiting of the phone first). Use Soft1 (Switch) to switch the two calls to talk. User can also use Soft1 (Conf) to make the second call when there is just an active call.

3.1.3. Ending a call

- Hangs up by handset onhook.

- Hangs up by press when in hands-free.

- Hangs up a call in call waiting state.

When there are two calls，user might use Soft1(Switch)to switch to the call you want to hang up first. Then press # or press Soft3(Close) key to finish talk, and phone will switch to the other call automatically.

Note：it is no use to press # to finish talk, if there is only one current call.

3.1.4. Transferring a call

- Blind Transfer

During talk, press

，and then dial the number that you want to transfer to, and press #. Phone will transfer the current call to the third party. After finishing transfer, the call you talk to will be hanged up. User can not select SIP line when phone transfers call.

- Attended Transfer

During talk, press

and input the number that you want to transfer to and press Soft2 (Send).

After that third party answers, then press

to complete the transfer. (You need enable call waiting and call transfer first). If there are two calls, you can just talk to one, and keep hold to the

other one. The one who is keep hold can not speak to you or hear from you. In this status, user can press * or Soft2 (Conf) to make calls mode in conference mode. If user wants to stop conference, user can press Soft1 (Split). (User must enable call waiting and three way call first).
Note: the server that user uses must support RFC3515 or it might not be used.

- Alert Transfer

During talk, press

firstly, then press Soft2(Send) after inputting the number that you want to transfer. You are waiting for connection, now, press Soft2(Transf) and the transfer will be done. (To use this feature, you need enable call waiting and call transfer first)

3.1.5. Calling Hold

During talking, user could press

to hold the current call. Press

again to unhold the call or switch the call active. This feature is also available in 3-way conference call.

3.1.6. 3-way conference call

User can press Soft1 (Conf) to dial the line2 (press Soft1(Answer) to answer the call directly if this call is from line2)during talking with line1. After line2 connect, user can press Soft2 (Conf) or * to enter into conference mode. To back to line1 from conference, please press Soft1 (Split); to end the call, please press Soft3 (End) or press

.

3.1.7. Switchboard Operator feature

User can press Soft1 (Conf) to dial the line2(press Soft1(Answer) to answer the call directly if this call is from line2) during talking with line1. After line2 connect, user can press Soft1 (Switch) to select which line you prefer to transfer, then press

to input the number you want to transfer and press

again to do the transfer.

3.1.8. Call records

TVP-2208P supports 100 items of missed call, 100 items of incoming call, and 100 items of outgoing call. If the records are full, the newest will replace the oldest. If phone's power cut or reboot, call records will be discarded.

- Missed call

Press

, and screen displays "Missed Call". Press Soft2 (Enter), phone will show the number and time of missed call. User can also use
to browse the missed call records, or press Soft2 (Detail) to check the details of this record, then press Soft2 (Dial) again to change the current number. Pressing Soft2(Dial) will call this number directly if user don't modify the number. If there is no missed call, screen will show "List Is Empty".

- Incoming call

Press

and switch the menu to "Incoming Call" by pressing
. Press Soft2 (Enter), phone will show the number of incoming call. User can also use
to browse the incoming call records; or press Soft2 (Detail) to check the details of this record, then press Soft2 (Dial) again to change the current number. Pressing Soft2 (Dial) will call this number directly if user don't modify the number. If there is no incoming call， screen will show "List Is Empty".

- Outgoing call

Press

, and use
to select to "Outgoing Call". Press Soft2 (Enter), phone will

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show the number and time of dialed call. User can also use to browse the dialed call records; or press Soft2 (Detail) to check the details of this record, then press Soft2 (Dial) again to change the current number. Pressing Soft2 (Dial) will call this number directly if user don't modify the number. If there is no dialed call, screen will show "List Is Empty". User can also press

to check "Outgoing Call".

3.2. The high-level operation

This VoIP Phone provides more advanced functions after setting at the permission scope of SIP server.

3.2.1. SMS function

- **Send message**

The followings list several methods to send message:

1.Press

in standby, then press Soft1(new) key. After inputting SMS content, press Soft2(send)key to input callee's number, next, press Soft2 again to send SMS.

2. Press

in standby, then press soft1(new) key. After inputting SMS content, press soft2(send) key, then pbook key to select your number to send SMS.

3. After inputting SMS content, user can press soft2(send) key, then input " #" and "the callee's IP address"to send SMS.

- **Browse Message and reply message**

When there's new message, phone will ring and remind by a small envelope on top of the screen, then press

, and Soft2(Enter) key to browse current new message. when there are more new messages come in, user can choose by using up and down keys, then press Soft2(Enter) key to check the sender's number and message content, next, press Soft2(Reply)key and input message content, finally, press Soft2(Send) again to reply this message.

Note: while user browses the message numbers, new messages will be marked by "new"; when user edits message, press # key that to switch input method, e.g. ABC (uppercase English input), abc (lowercase English input), 123(digit input), Korean (Korean input(if your phone's firmware version supports Korean). PY,(if your phone's firmware version supports Chinese)

3.2.2. Memo function

Press soft3 (Memo) key in standby, then Soft1(ADD) key, at this time, user can configure the future date time in terms of Time format, next, press down key to input the memo content, also can press # to switch input method, down key again to enter into reminder ring tone and down key at the third time to enter into ring mode. You can press right or left key to select your reminder ring tone after you enter into reminder ring tone, and select your ring mode by pressing right or left key after entering into ring mode. There are two ring modes, ring and text. Ring is reminder you by ring tone, text only show memo content without ring tone reminder. Finally, press soft2(save) key to save your memo.

Note: if there is memo notice when your phone is in call/off-hook/hands-free status, phone does not reminder by ring tone, only shows memo content in screen.

3.2.3. SpeedDial function

User can pre-defined numbers in these keys(numeric key 0-9). Hook off, press the defined numeric key, then input "#". Your pre-defined numbers will send out.

3.2.4. Realize Secondary Dial by Dialing for only one time

When you make secondary dial in off-hook/hands-free or standby pre-input mode, press

button to postpone input, and screen display will show ^. one stands for 2 seconds. For example, you input 123^45, the phone will send DTMF(45) 2 seconds after the phone call 123. 123^^^45 will make phone send DTMF(45) at 6 seconds interval

3.2.5. Phonebook prefix function

At standby mode, press phonebook button, user can not only select his needed number to call out but also he can add prefix to numbers, then call out. It is convenient for user add prefix numbers that PBX need.

3.2.6. Function key

If function key is set as SIP Line key, user can select which lines will be used to make call when dialing or make a 2nd dialing by this function key. Note that only the key which is registered is available to be select to call.

This function key can be configured as “Key Event”, namely set as F_MWI. It can set relative keys as Voice mail key, can check new and old voice mail; also can be set other function keys like the following table:

Field Name	Explanation
F_PBOOK	Like the phonebook key.
F_REDIAL	Like the redial key.
F_A_TRANSFER or F_B_TRANSFER	Like the transfer key.
F_PICKUP	Pickup function
F_JOIN	Joincall function.
F_AUTOREDIAL	Auto redial function.
F_UNAUTOREDIAL	Cancel redial function.
F_DND	Do not disturb function.
F_MWI	MWI(message waiting indication) function.
F_CFWD	Call forward function.
F_CALLERS	Like the Callers key.
F_MEMO	Like the Memo key.
F_REC	Record function (record on server).

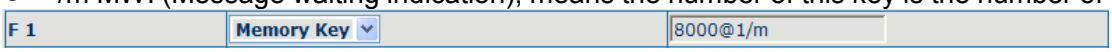
User can implement BLF/PRESENCE/MWI/SPEED DIAL features by Memory Key.

- /b Busy Lamp Field: Based on Asterisk, it can be used to check the status (Idle,ring,busy) of the pointed phones. It is helpful to operator to know the status of the phone which he will switch to.

User can configure the BLF like: 300 is rogatory number, @1 means SIP1, of course, user can configure as @2(SIP2); if don’t use this, simply says 300/b, it will use SIP1 as default. /b means use BLF feature.

When this configuration enable, the phone will subscribe the status of pointed phone each 60s: LED off means Idled, LED flash means ring and LED on means busy.

- /m MWI (Message waiting indication), means the number of this key is the number of voicemail



User can configure MWI function according to the above chart: 8000 is mailbox number, @1 is using SIP1, user also can configure @2(SIP2),the rest lines can be deduced by analogy, if no use, is 8000/m,it will pass the SIP1 line in default,/m means MIW function is using.

If there’s new voicemail, LED will blink and shows new message, after receiving, server will send current mail info to phone, after receiving new MWI order, LED will respond, if LED light is off, it means no new voicemail.

- /p Presence, means phone can check the status of other phone that has relevant numbers.

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F 1	Memory Key	500@1/p

User can configures presence function according to the above chart: 500 is number that search caller, @1 is using SIP1, user also can configure @2(SIP2),the rest lines can be deduced by analogy, if no use, is 500/p, it will pass the SIP1 line in default, /p means presence function is using. At this moment, press this button, it can show the correponding phone's status (on, off, fail,) which LED don't remind.

- /f speed dial, user configure it as same time as above attribute, after configuring, phone will implement above function in priority, then considering to perform speed dial
- /i PUSH TO TALK, user presses this button in standby, the phone can call other phone and the other phone will auto answer.

F 1	Memory Key	700/i

User can configure PUSH TO TALK according to the chart: 700 is number of callee.
After configuring, the phone can call 700 and make 700 auto answer by pressing this button.

3.2.7. Call pickup

Call pickup is implemented by simulating pickup function of PBX. it's that, when A calls B, B rings but no answer, at this moment, C can hook off and input an appointed prefix plus B's number, pick up A's call and talk with A.
The following chart shows how to configure an appointed prefix in dial peer to have call pick up function.

Number	Destination	Port	Mode	Alias	Suffix	Del Length
*1*T	0.0.0.0	5060	SIP	rep:pickup	no suffix	3

1 means appointed prefix code. After making the above configuration, C can dial *1* plus B'phone number to pick up A's call. User can set prefix in random, in the case of no affecting current dialing rules.

3.2.8. join call

When B is calling C, A can join in the existing call by inputing an appointed prefix numbers plus B or C number, if B or C also supports join call

The following chart shows how to configure an appointed prefix in dialpeer to have join call function.

Number	Destination	Port	Mode	Alias	Suffix	Del Length
*2*T	0.0.0.0	5060	SIP	rep:joincall	no suffix	3

2 means appointed prefix code. After making the above configuration, A can dial *2* plus B or C number to join B and C's call, . User can set prefix in random, in the case of no affecting current dialing rules.

3.2.9. redial/unredial

If B is in busy line when A calls B, A will get notice: busy, please hang up. If A want to connect B as soon as B is in idle, he can use redial function at the moment and he can dials an appointed prefix number plus B's number to realize redial function.

What is redial function? A can't not build a call with B when B is in busy ,then A will subscribe B's calling mode at 60 second intervals. once B is available, A will get reminder of rings to hook off, while A hooks off, A will call B automatically. If at this time A is occupied temporarily and unwilling to contact B, A also can cancel the redial function by dialing an appointed prefix plus B's number before making the redial function.

Number	Destination	Port	Mode	Alias	Suffix	Del Length
*3*T	0.0.0.0	5060	SIP	rep:redial	no suffix	3
*4*T	0.0.0.0	5060	SIP	rep:unredial	no suffix	3

3 is appointed prefix code. After making the above configuration, A can dial
3 plus B'phone number to make the redial function.
4 is appointed prefix code. After configuration, A can dial *4* to cancel redial function.

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User can set prefix in random, in the case of no affecting current dialing rules.

3.2.10. click to dial

When user A browses in an appointed Web page, user A can click to call user B via a link (this link to user B), then user A's phone will ring, after A hooks off, the phone will dial to B.

4. Setting

4.1. Introduction of configuration

4.1.1. Ways to configure

TVP-2208P has three different ways to different users.

- Use phone keypad.
- Use web browser（recommendatory way）.
- Use telnet with CLI command.

4.1.2. Password Configuration

There are two levels to access to phone: root level and general level. User with root level can browse and set all configuration parameters, while user with general level can set all configuration parameters except SIP (1-2) or IAX2's that some parameters can not be changed, such as server address and port. User will has different access level with different username and password.

- Default user with general level：
  - username：guest
  - password：guest
- Default user with root level：
  - username：admin
  - password：admin

The default password of phone screen menu is 123.

4.2. Setting via web browser

When this phone and PC are connected to network, enter the IP address of the wan port in this

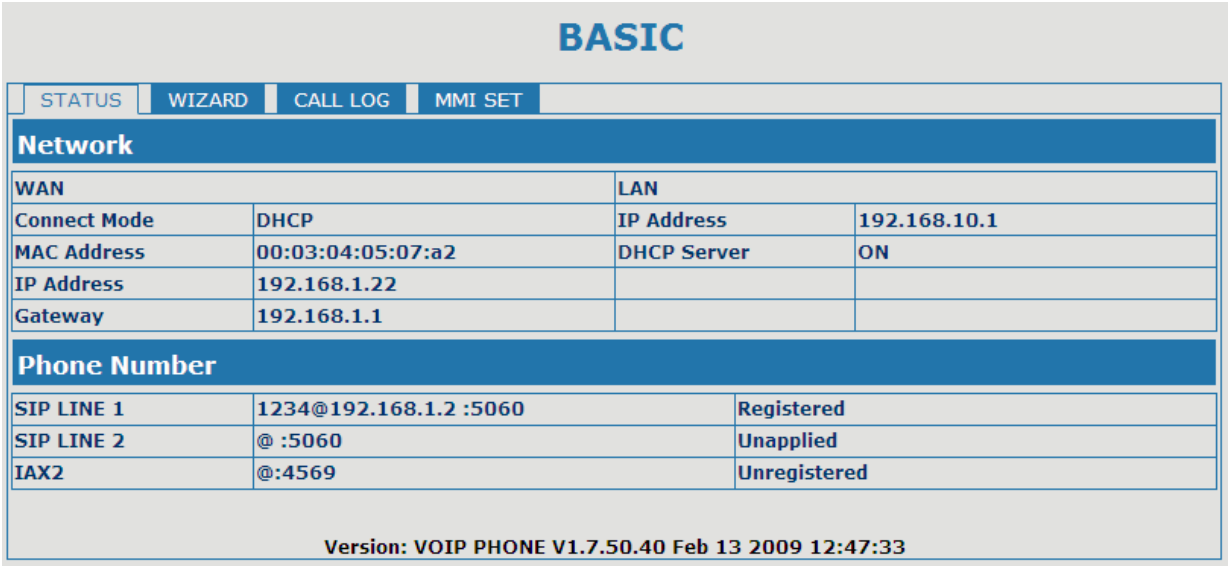
phone as the URL (e.g. http://xxx.xxx.xxx.xxx/ or http://xxx.xxx.xxx.xxx:xxxx/).
If you do not know the IP address, you can look it up on the phone's display by pressing button .
The login page is as below picture

※ :Input username and password, click "logon", and you will enter setting web interface.
There is a selection menu on the left side of the web interface. Click on the desired submenu; the current settings of this submenu will be displayed in the larger field on the right. You can now modify and store the values by using mouse and keyboard of your PC. To save the changes, click on the submenu "maintenance" and then click the "config" button and the "Save" button on the right field.

4.3. Configuration via WEB

4.3.1. BASIC

4.3.1.1. Status



Status

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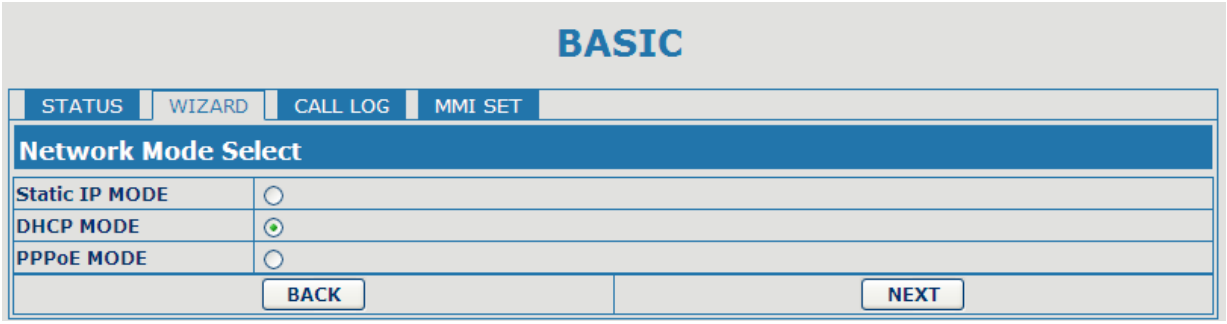
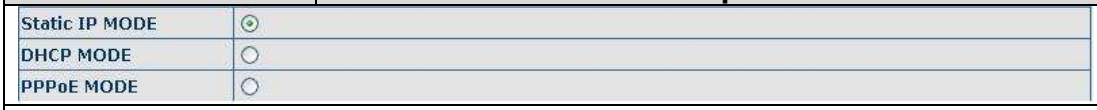
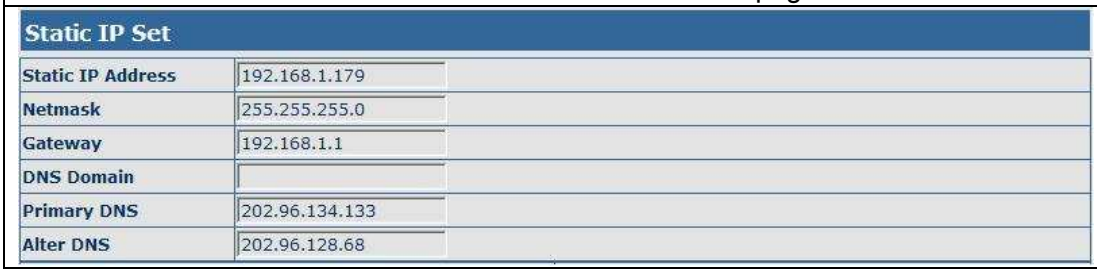
Field name	Explanation
Network	Shows the configuration information on WAN and LAN port, including the connect mode of WAN port (Static, DHCP, PPPoE), MAC address, the IP address of WAN port and LAN port, ON or OFF of DHCP mode of LAN port.
Phone Number	Shows the phone numbers provided by the SIP LINE 1-2 servers. The last line shows the version number and issued date.

4.3.1.2. Wizard



Wizard	
Field Name	**Explanation**
	
Please select the proper network mode according to the network condition. TVP-2208P provide three different network settings: ● Static: If your ISP server provides you the static IP address, please select this mode, then finish Static Mode setting. If you don't know about parameters of Static Mode setting, please ask your ISP for them. ● DHCP: In this mode, you will get the information from the DHCP server automatically; need not to input this information artificially. ● PPPoE: In this mode, your must input your ADSL account and password. You can also refer to 2.2. Initial Setting to speed setting your network.	
Choose Static IP MODE，click【**NEXT**】can config the network and SIP(default SIP1)easily, also can browse them too. Click【**BACK**】can return to the last page.	
	
Static IP Address	Input the IP address distributed to you.
Netmask	Input the Netmask distributed to you.
Gateway	Input the Gateway address distributed to you.
DNS Domain	Set DNS domain postfix. When the domain which you inputted can not be parsed, phone will automatically add this domain to the end of the domain which you inputted before and parse it again.
Primary DNS	Input your primary DNS server address.
Alter DNS	Input your standby DNS server address.

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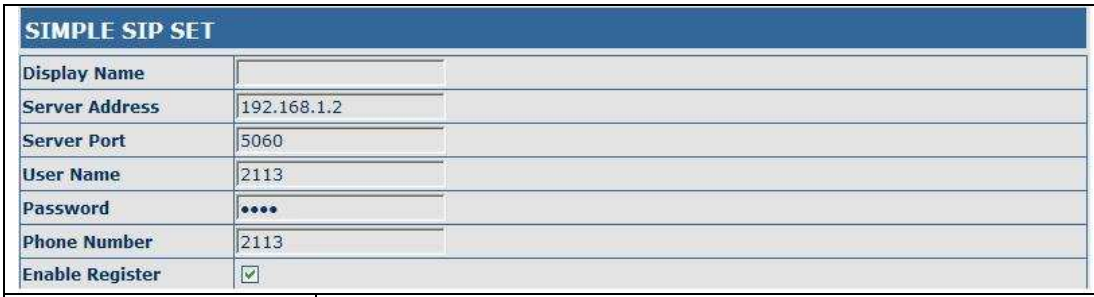
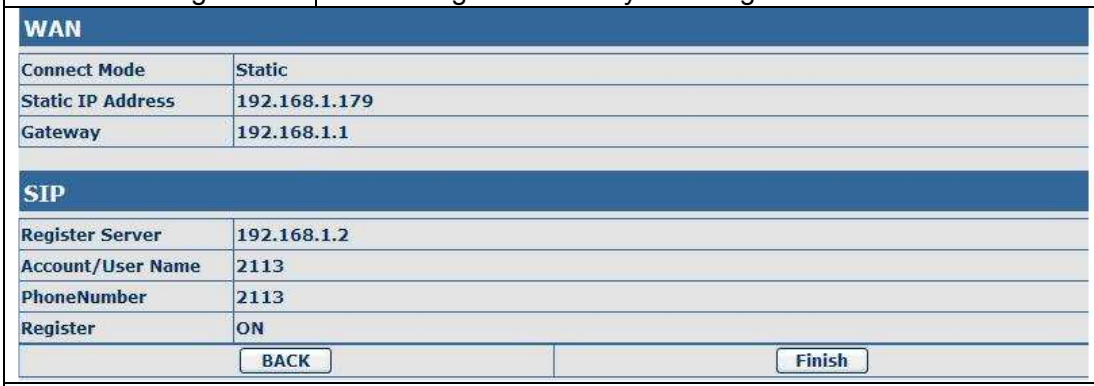
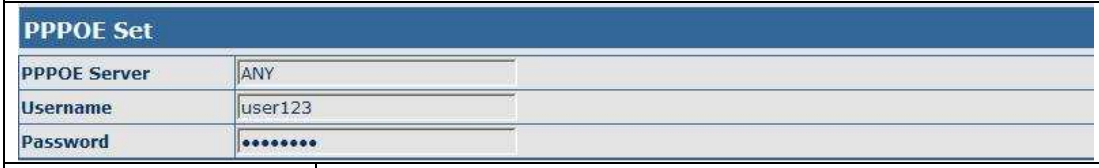


Display Name	If user set the display name, callee will show this display name.
Server Address	Input your SIP server address.
Server Port	Set your SIP server port.
User Name	Input your SIP register account name.
Password	Input your SIP register password.
Phone Number	Input the phone number assigned by your VOIP service provider.
Enable Register	Start to register or not by selecting it or not.



Display detailed information that you manual config.
Choose DHCP MODE，click【**NEXT**】to config simple SIP(default SIP1). You can browse it too. Click【**BACK**】to return to the last page. Like Static IP MODE。
Choose PPPoE MODE，click【**NEXT**】to config the PPPoE account/password and SIP(default SIP1). You can browse it too. Click【**BACK**】to return to the last page. Like Static IP MODE。

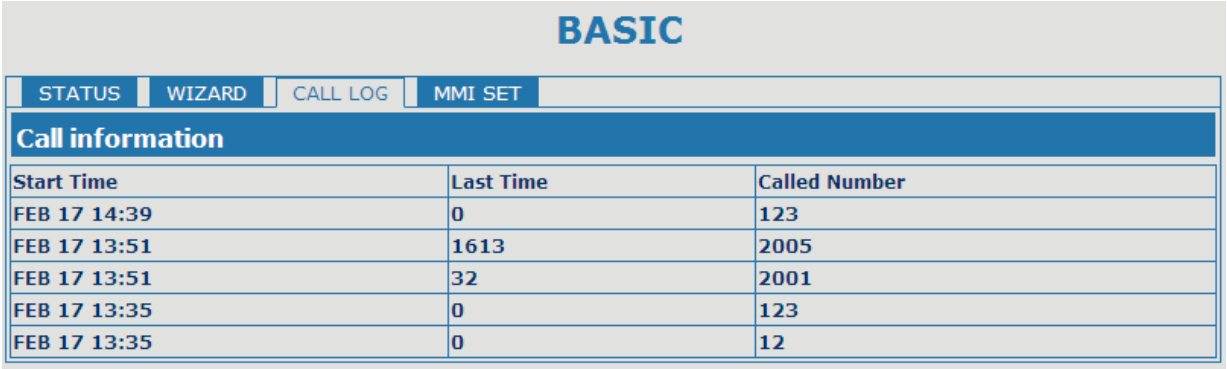


PPPoE Server	It will be provided by ISP.
Username	Input your ADSL account.
Password	Input your ADSL password.

Notice: Click【**Finish**】button after finish your setting, IP Phone will save the setting automatically and reboot. After reboot, you can dial by the SIP account.

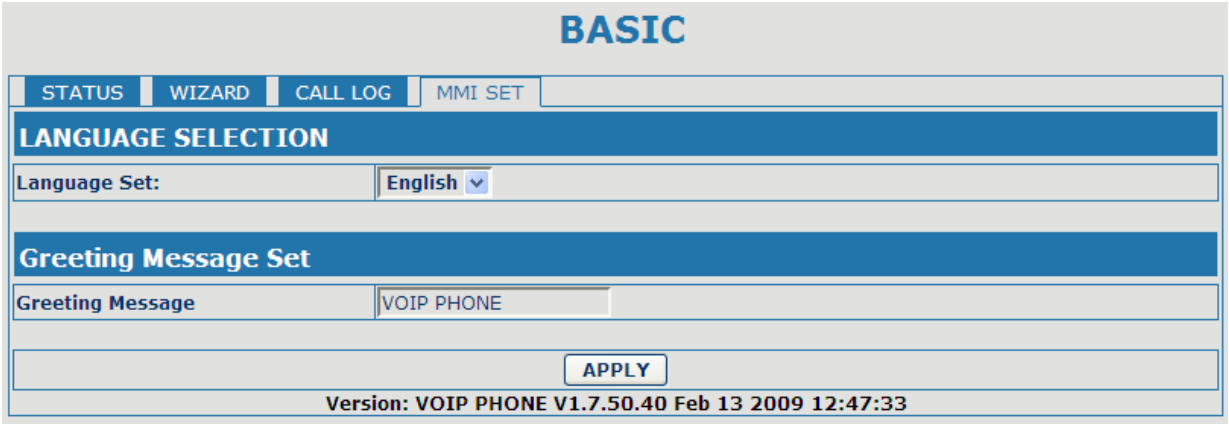
4.3.1.3. Call Log

You can look up all the outgoing calls through this page.



Call Log	
Field name	**explanation**
Start Time	Display the start time of the outgoing call
Last Time	Display the conversation time of the outgoing call.
Called Number	Display the account/protocol/line of the outgoing call.

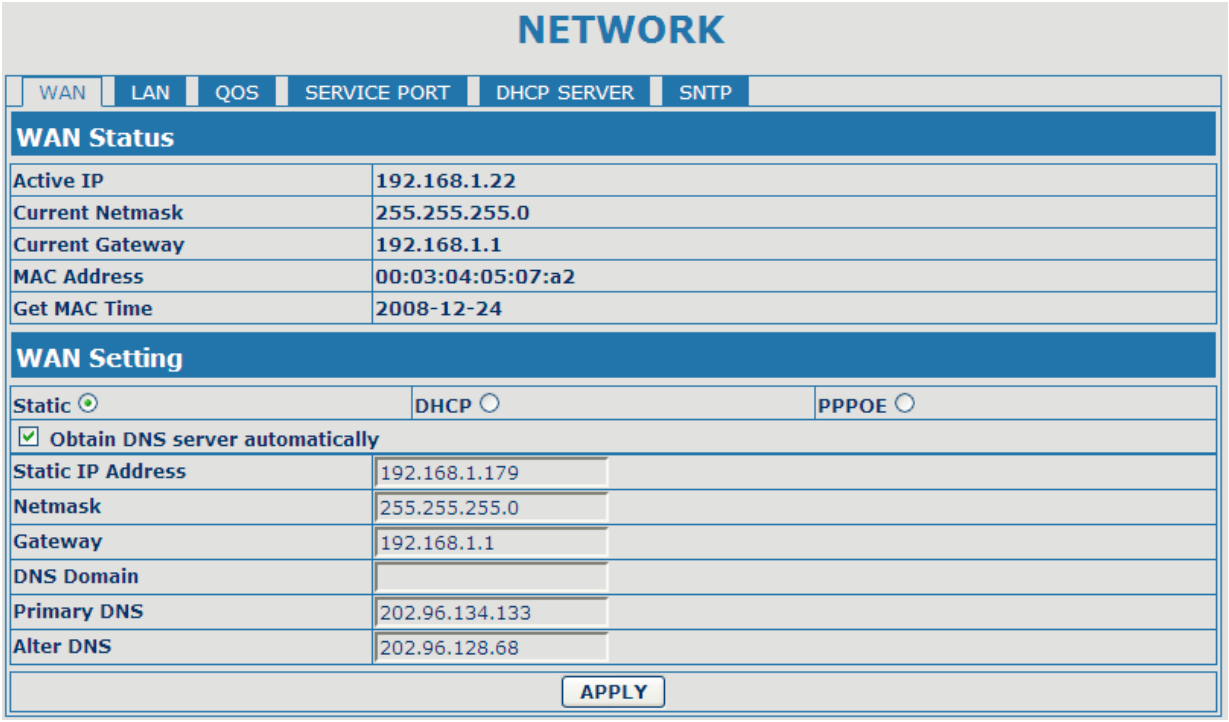
4.3.1.4. MMI SET



MMI SET	
Field name	**explanation**
Language Set	Set the language of phone, English is default.
Greeting Message	The greeting message will display on lcd when phone is idle. It can support 16 chars. the default chars are VOIP PHONE.

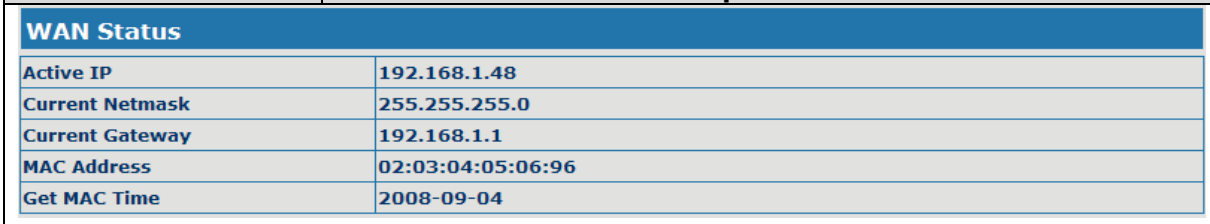
4.3.2. Network

4.3.2.1. WAN Config



WAN Config

Field Name	explanation



Active IP	The current IP address of the phone.
Current Netmask	The current Netmask address.
MAC Address	The current MAC address of the phone.
Current Gateway	The current Gateway IP address.
Get MAC Time	Shows the time of getting MAC address

Please select the proper network mode according to the network condition. TVP-2208P provide three different network settings:

- Static: If your ISP server provides you the static IP address, please select this mode, then finish Static Mode setting. If you don't know about parameters of Static Mode setting, please ask your ISP for them.
- DHCP: In this mode, you will get the information from the DHCP server automatically; need not to input this information artificially.
- PPPoE: In this mode, your must input your ADSL account and password.

You can also refer to 2.2. Initial Setting to speed setting your network.

Obtain DNS server automatically	Using the DHCP mode to get the DNS address. If disable, it will using the DNS address in the Static mode, the default is enable.

Static IP Address	192.168.1.179
Netmask	255.255.255.0
Gateway	192.168.1.1
DNS Domain	
Primary DNS	202.96.134.133
Alter DNS	202.96.128.68

If you use static mode, you need set it.

IP Address	Input the IP address distributed to you.
Netmask	Input the Netmask distributed to you.
Gateway	Input the Gateway address distributed to you.
DNS Domain	Set DNS domain postfix. When the domain which you inputted can not be parsed, phone will automatically add this domain to the end of the domain which you inputted before and parse it again.
Primary DNS	Input your primary DNS server address.
Alter DNS	Input your standby DNS server address.

PPPOE Server	ANY
Username	user123
Password	••••••••

If you uses PPPoE mode， you need to make the above setting.

PPPoE Server	It will be provided by ISP.
Username	Input your ADSL account.
Password	Input your ADSL password.

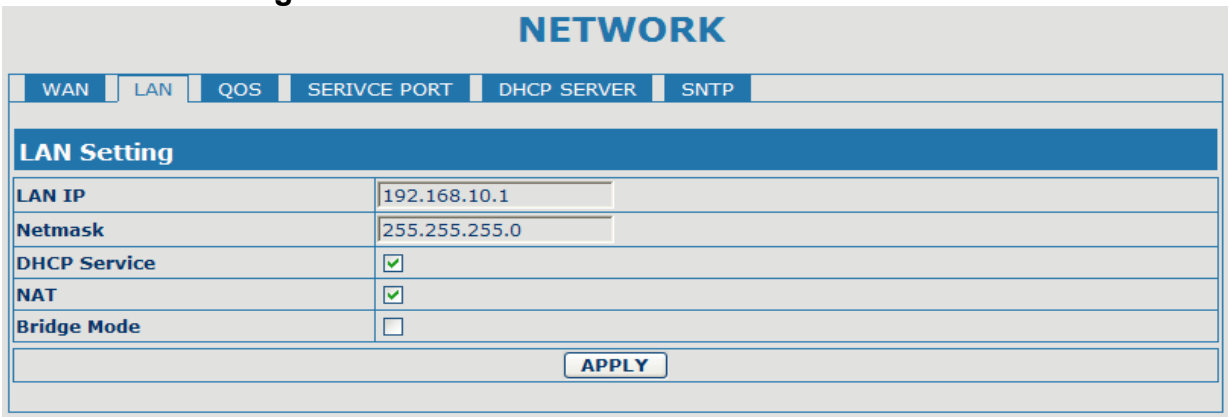
Notice:

1）Click “Apply” button after finishe your setting, IP Phone will save the setting automatically and new setting will take effect.

2）If you modify IP address, the web will not response by the old IP address. Your need input new IP address in the address column to logon in the phone.

3）If networks ID which is distributed by DHCP server is same as network ID which is used by LAN of system, phone will use the DHCP IP to set WAN, and modify LAN’s networks ID(for example, system will change LAN IP from 192.168.10.1 to 192.168.11.1) when phone uses DHCP client to get IP in startup; if phone uses DHCP client to get IP in running status and network ID is also same as LAN’s, phone will refuse to accept the IP to configure WAN.

4.3.2.2. LAN Config



LAN Config	
Field name	**explanation**
LAN IP	Specify LAN static IP.
Netmask	Specify LAN Netmask.
DHCP Service	Select the DHCP server of LAN port or not. After user modify the LAN IP address, phone will amend and adjust the DHCP Lease Table and

	save the result amended automatically according to the IP address and Netmask. You need restart the phone and the DHCP server setting will take effect.
NAT	Select NAT or not.
Bridge Mode	Select Bridge Mode or not: If you select Bridge Mode, the phone will no longer set IP address for LAN physical port，LAN and WAN will join in the same network.. Click “Apply”, the phone will reboot.
Notice: If you choose the bridge mode, the LAN configuration will be disabled.	

4.3.2.3. Qos Config

The VOIP phone support 802.1Q/P protocol and DiffServ configuration. VLAN functionality can use different VLAN IDs by setting signal/voice VLAN and data VLAN. The VLAN application of this phone is very flexible.

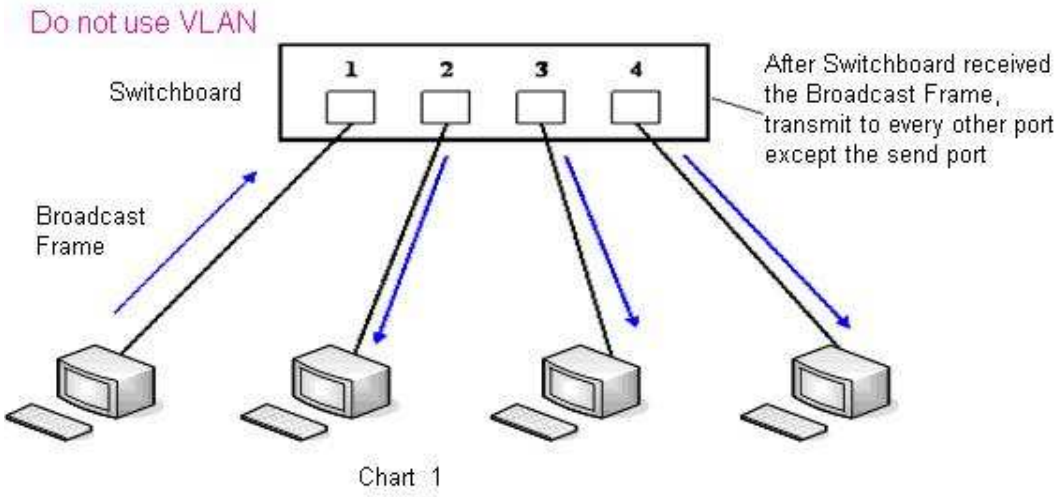


Chart 1

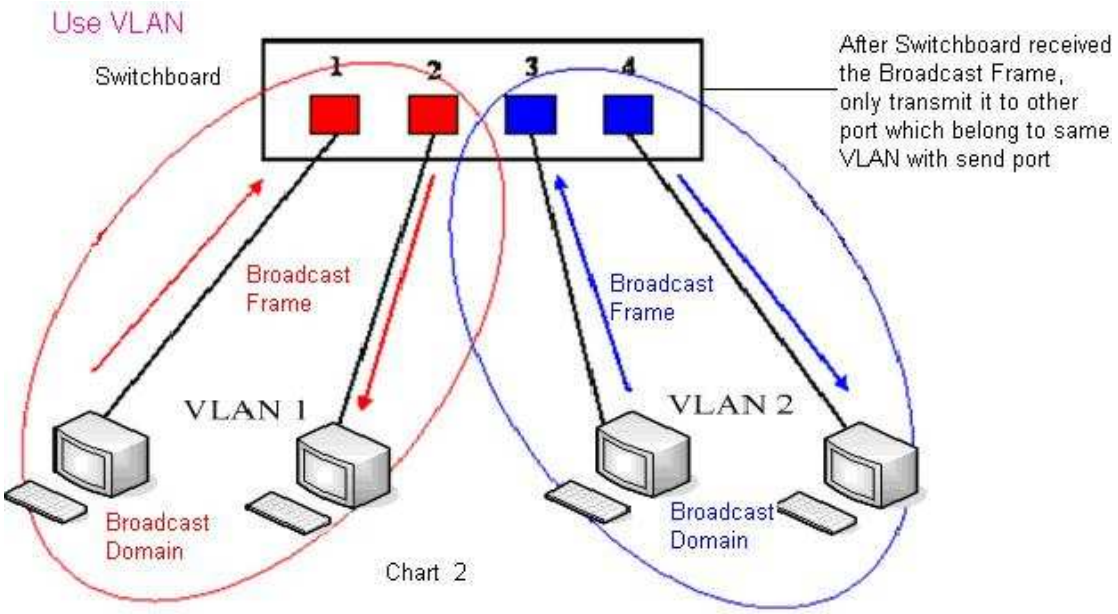


Chart 2

In chart 1, there is a layer 2 switch without setting VLAN. Any broadcast frame will be transmitted to the other ports except the send port. For example, a broadcast information is sent out from port 1 then transmitted to port 2,3and 4.

In chart 2, red and blue indicate two different VLANs in the switch, and port 1 and port 2 belong to red VLAN, port 3 and port 4 belong to blue VLAN. If a broadcast frame is sent out from port 1, switch will transmit it to port 2, the other port in the red VLAN and not transmit it to port3 and port 4 in blue VLAN. By this means, VLAN divide the broadcast domain via restricting the range of

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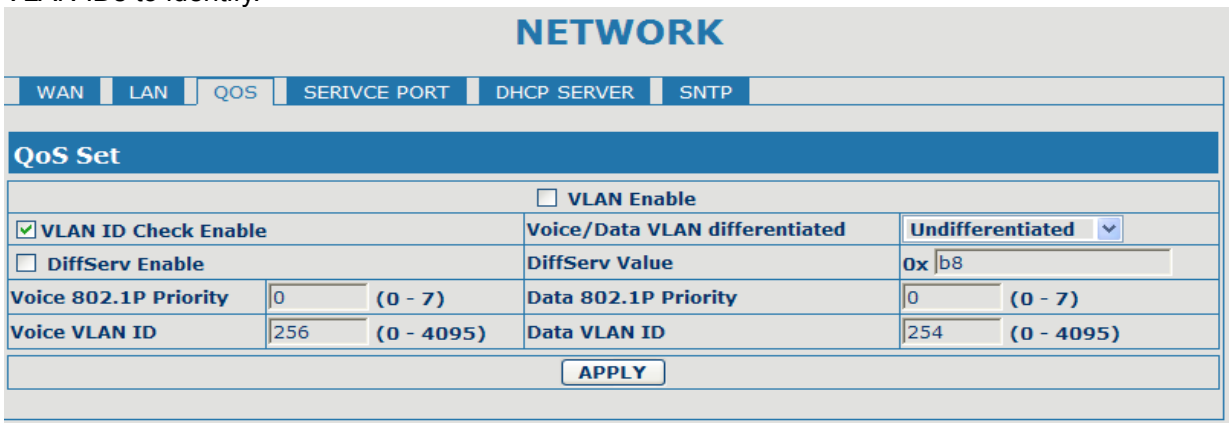
broadcast frame transmition.
Note: chart 2 use red and blue to identify the different VLAN, but in practice, VLAN uses different VLAN IDs to identify.



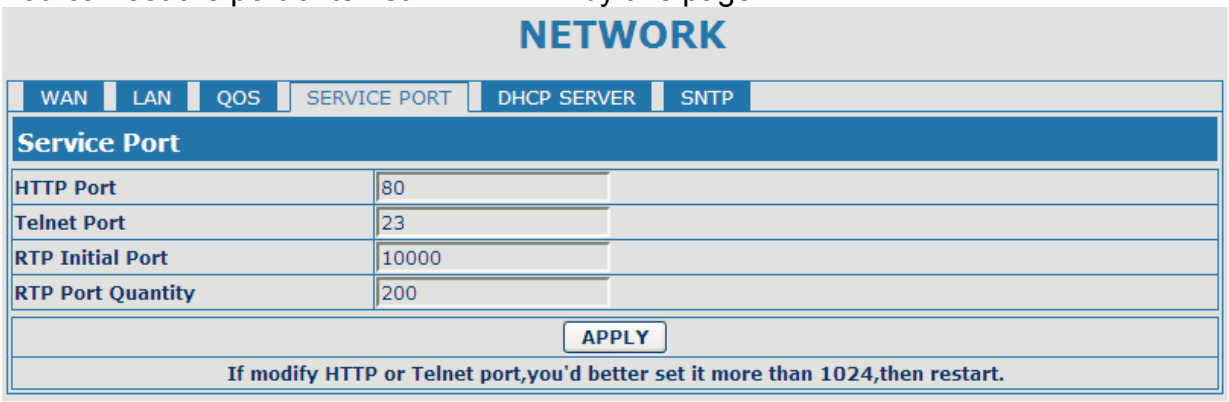
QoS Configuration	
Field name	**explanation**
VLAN Enable	Before select it to enable VLAN, you need enable Bridge mode in LAN config.
VLAN ID Check Enable	Enable VLAN ID check by selecting it. After enable VLAN ID check, if VLAN ID of a data package is not the same with the phone's or a data package do not have VLAN ID, the data package will be discarded.
Voice/Data VLAN differentiated	After enable VLAN, system will set packets with different type of VLAN ID. Undifferentiated means after using VLAN, both VoIP packets and other data packets will use the voice VLAN ID; tag differentiated means after using VLAN, VoIP(signal and voice) packets will add voice VLAN ID, and other data packets will add data VLAN ID; data untaged means after using VLAN, only VoIP packets will add voice VLAN ID. Other data packets will not use VLAN.
DiffServ Enable	Select it or not to Enable or disable DiffServ.
DiffServ Value	Set DiffServ value, the common value is 0x00.
Voice 802.1P Priority	Specify 802.1P Priority of voice/signal data package.
Data 802.1P Priority	Set 802.1p of data VLAN. Non-VoIP data (such as http, telnet, ping etc) will use this value to set VLAN package.
Voice VLAN ID	Set VLAN ID of voice/signal data package.
Data VLAN ID	Set 802.1q of data VLAN ID. Non-VoIP data (such as http, telnet, ping etc) will use this value to set VLAN package.

NOTICE：
1）Startup VLAN, if set Voice/Data VLAN differentiated as Undifferentiated, all packets will use the Voice VLAN ID as the tag.
2) Startup VLAN, if set Voice/Data VLAN differentiated as tag differentiated and disable the DiffServ, then system will not distinguish the voice and data, all packets will use the Voice VLAN ID as the tag.
3) Startup VLAN, if set Voice/Data VLAN differentiated as tag differentiated and enable the DiffServ, then system will distinguish the voice and data and add the VLAN ID each other.
4) Startup VLAN, if set Voice/Data VLAN differentiated as data untaged, then the packet of the signal/voice will use the Voice VLAN ID as the tag, but the data packets will not take the VLAN tag.
5) If Disable the VLAN, regardless to set the Voice/Data VLAN differentiated or not, all packets will not take the VLAN tag; If enable the DiffServ, all packets will only take the DiffServ value.
6) user need notice, enable the VLAN ID Check Enable that is default, If enable it, the phone will match the VLAN ID strictly. When others' VLAN ID dismatch with us, the packets will discard. Contrarily, the phone will accept the packets with the distinct VLAN ID.
7) You must gain the IP with the Static mode when you set VLAN, otherwise can't gain the IP in

the VLAN and also can not dial with point to point.

4.3.2.4. Service Port

You can set the port of telnet/HTTP/RTP by this page.



SERVICE PORT	
Field name	**explanation**
HTTP Port	set web browse port, the default is 80 port，if you want to enhance system safety，you'd better change it into non-80 standard port； Example: The IP address is 192.168.1.70. and the port value is 8090, the accessing address is http://192.168.1.70:8090
Telnet Port	Set Telnet Port, the default is 23. You can change the value into others. Example: The IP address is 192.168.1.70. the telnet port value is 8023, the accessing address is telnet 192.168.1.70 8023
RTP Initial Port	Set the RTP Initial Port. It is dynamic allocation.
RTP Port Quantity	Set the maximum quantity of RTP Port, the default is 200.
Notice: 1）You need save the configuration and reboot the phone after set this page. 2）If you modify the port of Telnet and HTTP, you would better set the value more than 1024 because the port value less than 1024 is system port reserved. 3）if you set 0 for the HTTP port, it will disable HTTP service.	

4.3.2.5. DHCP SERVER

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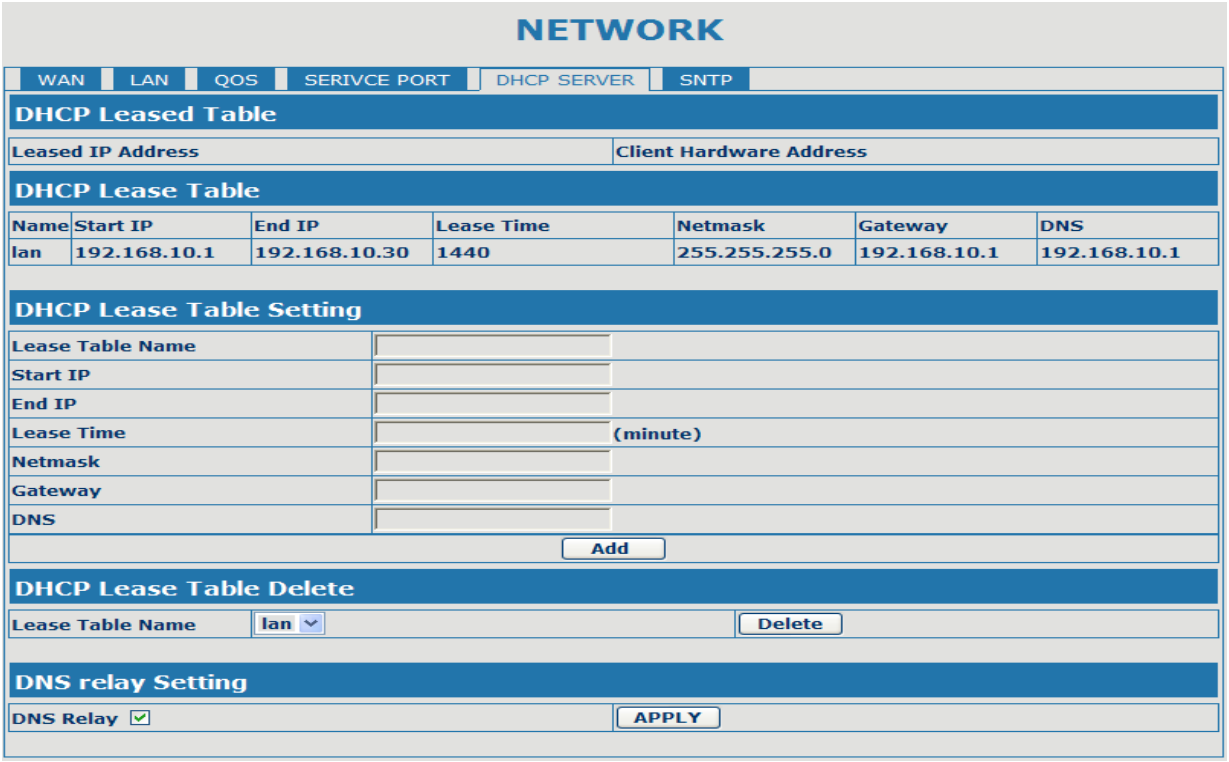


DHCP SERVER	
Field name	**explanation**
DHCP Leased Table	IP-MAC mapping table. If the LAN port of the phone connects to a device, this table will show the IP and MAC address of this device.

DHCP Lease Table

Name	Start IP	End IP	Lease Time	Netmask	Gateway	DNS
lan	192.168.10.1	192.168.10.30	1440	255.255.255.0	192.168.10.1	192.168.10.1

Shows the DHCP Lease Table， the unit of Lease time is Minute.	
Lease Table Name	Specify the name of the lease table
Start IP	Set the start IP address of the lease table
End IP	Set the end IP address of the lease table, the network device connected to LAN port will get IP address between Start IP and End IP by DHCP.
Netmask	Set the Netmask of the lease table
Gateway	Set the Gateway of the lease table
Lease Time	Set the Lease Time of the lease table
DNS	Set the default DNS server IP of the lease table; Click the **Add** button to submit and add this lease table

DHCP Lease Table Delete

Lease Table Name | lan | Delete

Select name of lease table, click the **Delete** button will delete the selected lease table from DHCP lease table.	
DNS Relay	Select DNS Relay, the default is enable. Click the Apply button to become effective.

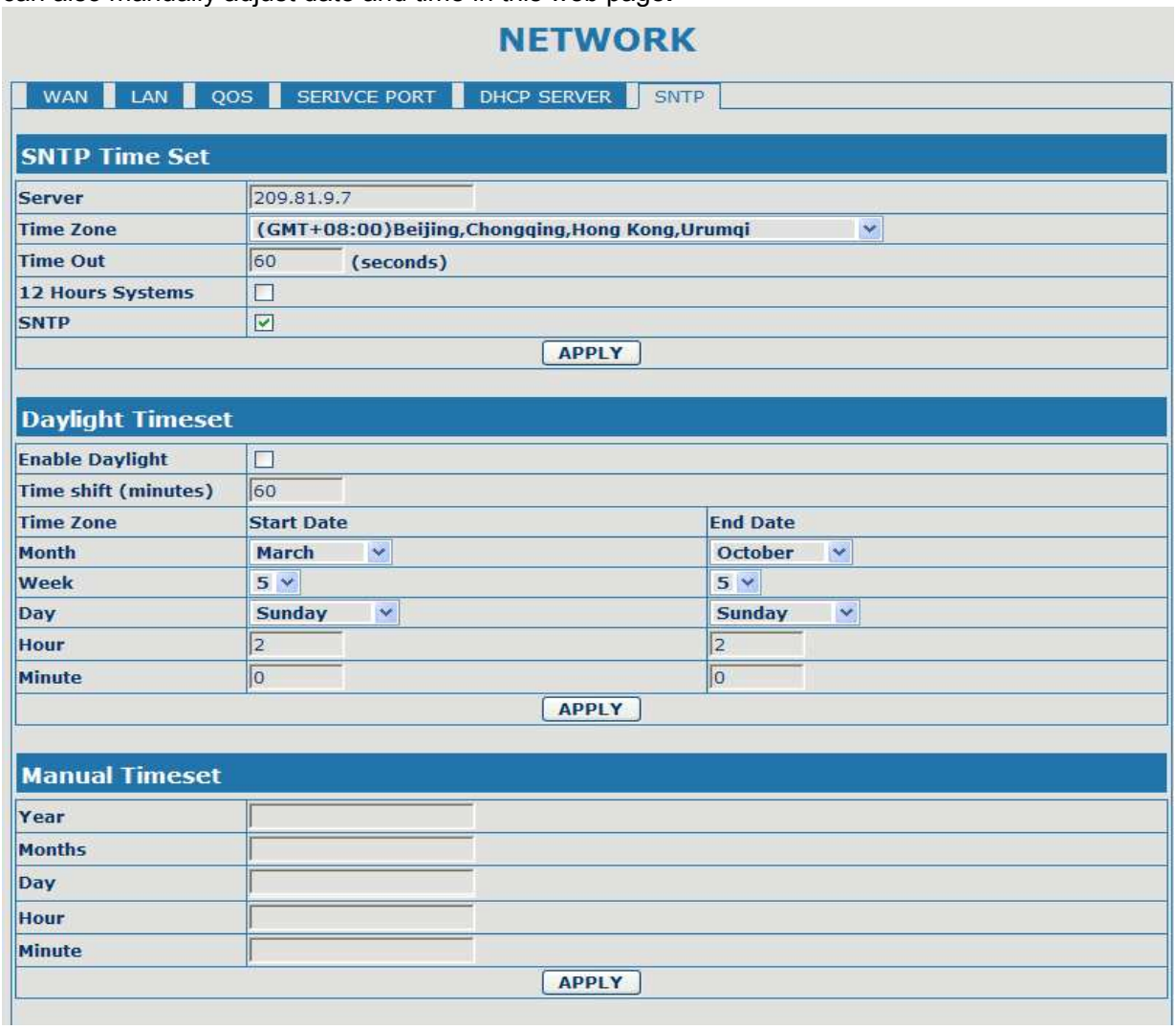
Notice:

1）The size of lease table can not be larger than the quantity of C network IP address. We recommend you to use the default lease table and not modify it.

2）If you modifies the DHCP lease table， you need save the configuration and reboot.

4.3.2.6. SNTP

Setting time zone and SNTP (Simple Network Time Protocol) server according to your location, you can also manually adjust date and time in this web page.



SNTP	
Field name	**explanation**
Server	Set SNTP Server IP address.
Time Zone	Select the Time zone according to your location.
Time Out	Set the time out, the default is 60 seconds.
12 Hours Systems	Swich the time mechanism between 12 hours and 24 hours. Default is 24 hours mode
SNTP	Select the SNTP, and click Apply to make the SNTP Times effective.
Enable Daylight	Enable daylight saving time
Time shift(minutes)	Setup the variety length
Month	Setup stat and end month
Week	Setup start and end week
Day	Setup start and end day
Hour	Setup start and end hours
Minute	Setup start and end minutes

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Year	
Months	
Day	
Hour	
Minute	
APPLY	

Notice: You need specify the above all items.

4.3.3. VOIP

4.3.3.1. SIP Config

Set your SIP server in the following interface.

SIP Config	
Field name	**explanation**

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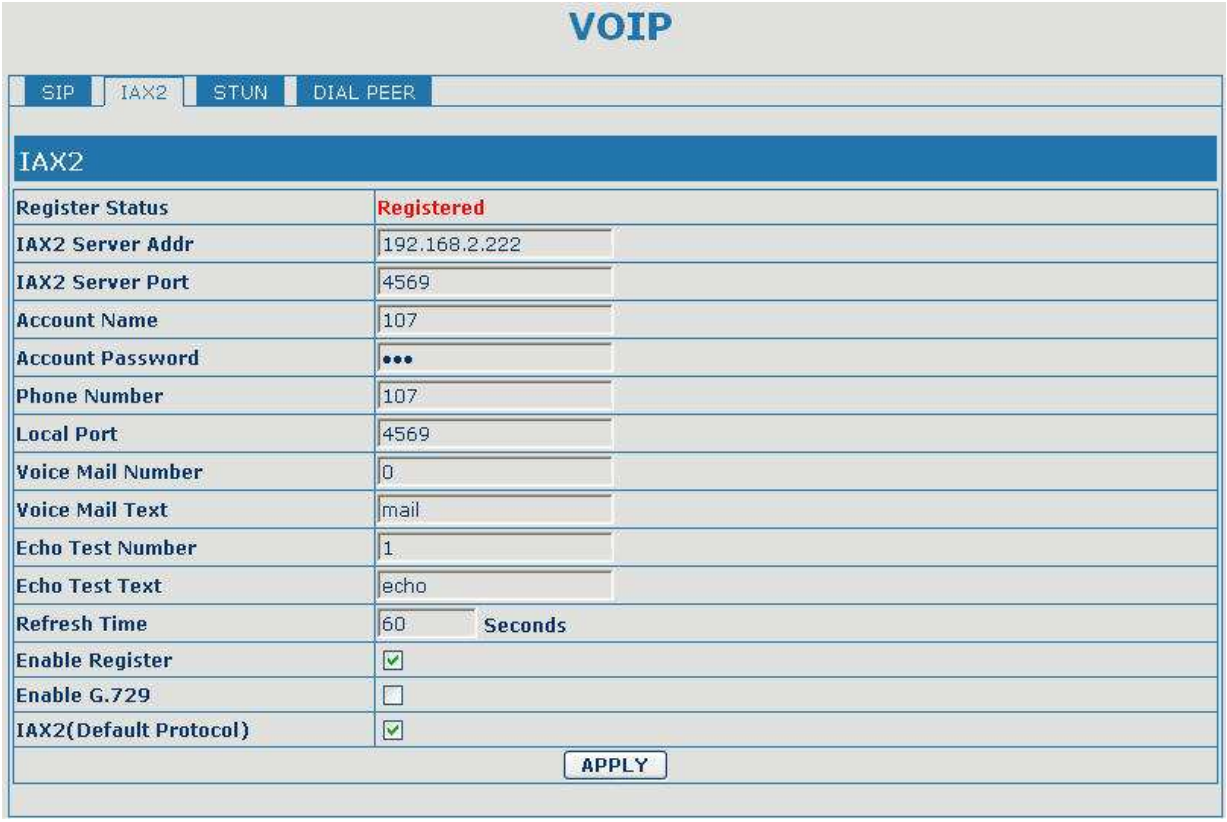
Choose line to set info about SIP, there are 2 lines to choose. You can switch by 【**Load**】 button.	
Register Status	Shows if the phone has been registered the SIP server or not; or so, show Unapplied;
Server Name	Set the server name.
Server Address	Input your SIP server address.
Server Port	Set your SIP server port.
Account Name	Input your SIP register account name.
Password	Input your SIP register password.
Phone Number	Input the phone number assigned by your VoIP service provider. Phone will not register if there is no phone number configured.
Display Name	Set the display name.
Proxy Server Address	Set proxy server IP address（Usually, Register SIP Server configuration is the same as Proxy SIP Server. But if your VoIP service provider give different configurations between Register SIP Server and Proxy SIP Server, you need make different settings.）
Proxy Server Port	Set your Proxy SIP server port.
Proxy Username	Input your Proxy SIP server account.
Proxy Password	Input your Proxy SIP server password.
Domain Realm	Set the sip domain if needed, otherwise this VoIP phone will use the Register server address as sip domain automatically. (Usually it is same with registered server and proxy server IP address).
Enable Register	Start to register or not by selecting it or not.
Register Expire Time	Set expire time of SIP server register, default is 60 seconds. If the register time of the server requested is longer or shorter than the expire time set, the phone will change automatically the time into the time recommended by the server, and register again.
NAT Keep Alive Interval	Set examining interval of the server, default is 60 seconds
User Agent	Set the user agent if have, the default is VoIP Phone 1.0
Signal Key	Set the key for signal encryption
Media Key	Set the key for RTP encryption
Local port	Set sip port of each line
Ring type	Set ring type of each line
Hot line Number	Set Hot line number of each line.
Transfer Expire Time	The phone send bye and end the call as soon as hang up.
Enable Subscribe	Enable Subscribe.
Enable Keep Authentication	Enable/Disable Keep Authentication.
NAT Keep Alive	Enable/Disable keeps NAT of SIP alive. If some server refuse to register with too short interval time, and has no packets sending to device in private network to keep NAT alive, user could set this function ON. It need set the keep alive interval time less than the NAT server's.
Enable Via rport	Enable/Disable system to support RFC3581. Via rport is special way to realize SIP NAT.
Enable PRACK	Enable or disable SIP PRACK function, suggest use the default config.
Long Contact	Set more parameters in contact field; connection with SEM server
Enable URI Convert	Convert # to %23 when send the URI.
Dial Without Register	Set call out by proxy without registration;
Ban Anonymous Call	Set to ban Anonymous Call;
Forward Type	Select call forward mode, the default is Off ● Off：Close down calling forward ● Busy：If the phone is busy, incoming calls will be forwarded to the

	appointed phone. ● No answer： If there is no answer, incoming calls will be forwarded to the appointed phone. ● Always：Incoming calls will be forwarded to the appoint phone directly. The phone will Prompt the incoming while doing forward.
Forward Phone Number	Appoint your forward phone number.
Server Type	Select the special type of server which is encrypted, or has some unique requirements or call flows.
DTMF Mode	Select DTMF sending mode, there are three modes: ● DTMF_RELAY ● DTMF_RFC2833 ● DTMF_SIP_INFO Different VoIP Service providers may provide different modes.
RFC Protocol Edition	Select SIP protocol version to adapt for the SIP server which uses the same version as you select. For example, if the server is CISCO5300, you need to change to RFC2543, else phone may not cancel call normally. System uses RFC3261 as default.
Transport Protocol	Set transport protocols, TCP or UDP;
RFC Privacy Edition	Set Anonymous call out safely; Support RFC3323and RFC3325;
Subscribe Expire Time	Set the interval of Subscribe.
Enable DNS SRV	Support DNS looking up with _sip.udp mode
Click to Talk	Set click to Talk (need practical software support).
Signal Encode	Enable/Disable Signal Encrypt.
RTP Encode	Enable/Disable RTP Encrypt.
Enable Session Timer	Set Enable/Disable Session Timer, whether support RFC4028.It will refresh the SIP sessions.
Answer With Single Codec	Enable/Disable the function when call is incoming, phone replies SIP message with just one codec which phone supports.
Auto TCP	Set to use automatically TCP protocol to guarantee usability of transport as message is above 1300 byte
Enable Strict Proxy	Support the special SIP server-when phone recieves the patckets sent from server， phone will use the source IP address, not the address in via field.
Enable GRUU	Set to support GRUU
Enable Displayname Quote	Set to make quotation mark to displayname as the phone sends out signal, in order to be compatible with server.

4.3.3.2. IAX2 Config



IAX2 Config	
Field name	**explanation**
Register Status	Shows if the phone has been registered the IAX2 server or not.
IAX2 Server Addr	Input your IAX2 server address.
IAX2 Server Port	Set your IAX2 server port, the default is 4569.
Account Name	Input your IAX2 register account name.
Account Password	Input your IAX2 register password.
Phone Number	Input your assigned phone number (usually it is same you're your IAX2 account name).
Local Port	Set your local sport，the default is 4569.
Voice Mail Number	Specify the voice mail's number.
Voice Mail Text	Specify the voice mail's name.
Echo Test Number	Set echo test number. If IAX2 server supports echo test, and echo test number is non- numeric, system could set an echo test number to replace the echo test text. So user can dial the numeric number to test echo voice test. This function is provided with server to make endpoint to test whether endpoint could talk through server normally.
Echo Test Text	Specify echo test text's name.
Refresh Time	Set expire time of IAX2 server register, you can set it between 60 and 3600 seconds.
Enable Register	Start to register the IAX2 server or not by selecting it or not.
Enable G.729	Enable or disable code G.729 by selecting it or not
IAX2(Default Protocol)	Enable or disable IAX2 as default dial protocol

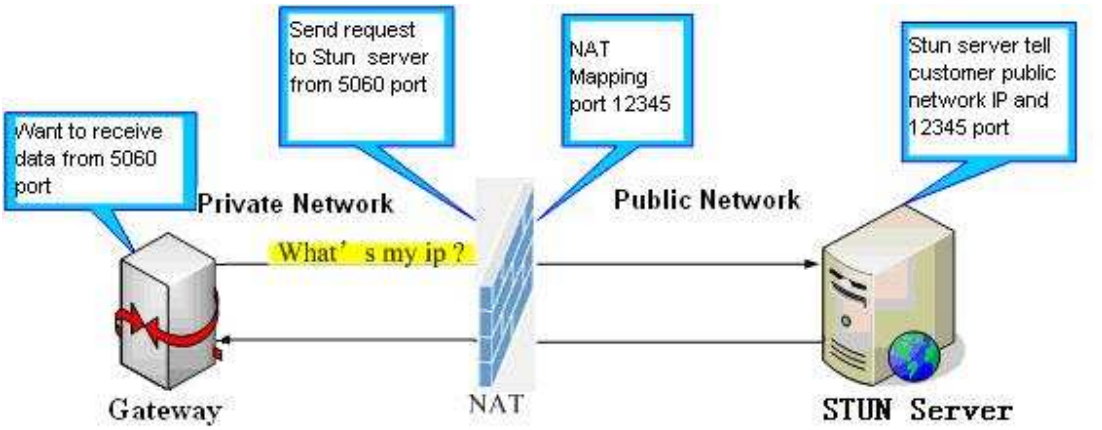
4.3.3.3. STUN Config

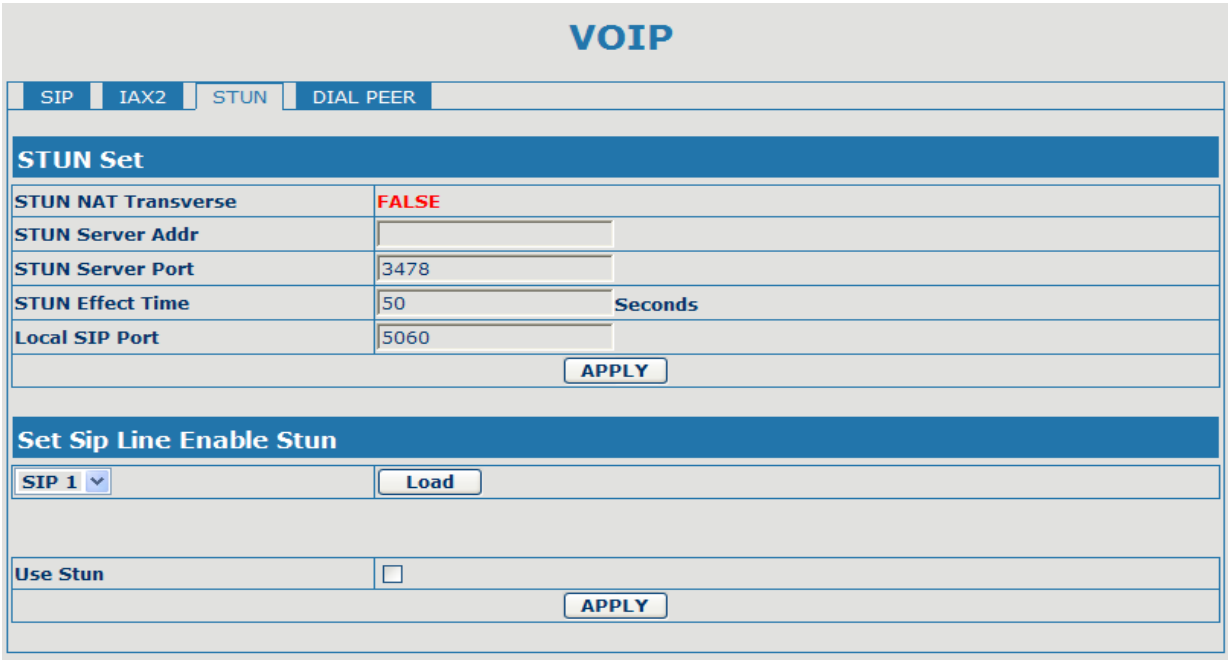
In this web page, you can config SIP STUN.

STUN:

By STUN server, the phone in private network could know the type of NAT and the NAT

mapping IP and port of SIP. The phone might register itself to SIP server with global IP and port to realize the device both calling and being called in private network.


Send request to Stun server from 5060 port
NAT Mapping port 12345
Stun server tell customer public network IP and 12345 port
Want to receive data from 5060 port
Private Network
Public Network
What' s my ip ?
Gateway
NAT
STUN Server



STUN	
Field name	**explanation**
STUN NAT Transverse	Shows STUN NAT Transverse estimation, true means STUN can penetrate NAT, while False means not.
STUN Server Addr	Set your SIP STUN Server IP address
STUN Server Port	Set your SIP STUN Server Port
STUN Effect Time	Set STUN Effective Time. If NAT server finds that a NAT mapping is idle after time out, it will release the mapping and the system need send a STUN packet to keep the mapping effective and alive.
Local SIP Port	Set the SIP port.
Set Sip Line Enable Stun SIP 1 Load	
Choose line to set info about SIP, There are 2 lines to choose. You can switch by 【**Load**】 button.	
Use Stun	Enable/Disable SIP STUN.

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Notice: SIP STUN is used to realize SIP penetration to NAT. If your phone configures STUN Server IP and Port (default is 3478), and enable SIP Stun, you can use the ordinary SIP Server to realize penetration to NAT.

4.3.3.4. DIAL PEER setting

This functionality offers you more flexible dial rule, you can refer to the following content to know how to use this dial rule. When you want to dial an IP address, the entry of IP addresses is very cumbersome, but by this functionality, you can set number 156 to replace 192.168.1.119 here.

Number	Destination	Port	Mode	Alias	Suffix	Del Length
156	192.168.1.119	5060	SIP	no alias	no suffix	0

When you want to dial a long distance call to Beijing, you need dial an area code 010 before local phone number, but you can also dial number 1 instead of 010 after we make a setting according to this dial rule. For example, you want to dial 01062213123, but you need dial only 162213123 to realize your long distance call after you make this setting.

Number	Destination	Port	Mode	Alias	Suffix	Del Length
1T	0.0.0.0	5060	SIP	rep:010	no suffix	1

To save the memory and avoid abundant input of user,add the follow fuctions:

Number	Destination	Port	Mode	Alias	Suffix	Del Length
13xxxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0
13[5-9]xxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0

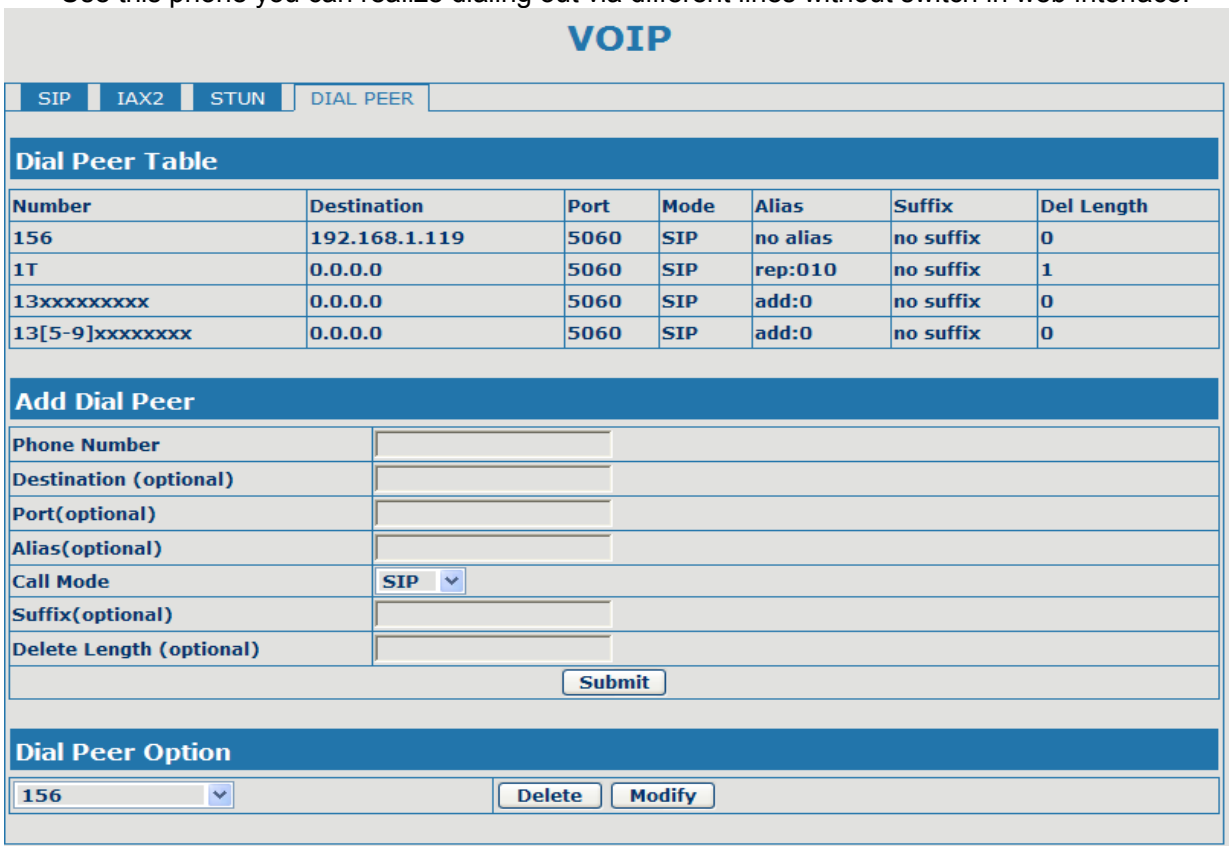
1、x Match any single digit that is dialed.

If user makes the above configuration, after user dials 11 digit numbers started with 13, the phone will send out 0 plus the dialed numbers automatically.

2、[] Specifies a range that will match digit. It may be a range, a list of ranges separated by commas, or a list of digits.

If user makes the above configuration, after user dials 11 digit numbers started with from 135 to 139, the phone will send out 0 plus the dialed numbers automatically.

Use this phone you can realize dialing out via different lines without switch in web interface.



DIAL PEER

Field name	explanation
Phone number	There are two types of matching conditions: one is full matching, the other is prefix matching. In the Full matching, you need input your desired phone number in this blank, and then you need dial the phone number to realize calling to what the phone number is mapped. In the prefix matching, you need input your desired prefix number and T; then dial the prefix and a phone number to realize calling to what your prefix number is mapped. The prefix number supports at most 30 digits
Destination	Set Destination address. This is optional config item. If you want to set peer to peer call, please input destination IP address or domain name. If you want to use this dial rule in SIP2 line, you need input 255.255.255.255 or 0.0.0.2 in it.
Port	Set the Signal port, the default is 5060 for SIP.
Alias	Set alias. This is optional config item. If you don't set Alias, it will show no alias.
Note: There are four types of aliases. 1) add: xxx, it means that you need dial xxx in front of phone number, which will reduce dialing number length. 2) all: xxx, it means that xxx will replace some phone number. 3) del: It means that phone will delete the number with length appointed. 4) Rep: It means that phone will replace the number with length and number appointed. You can refer to the following examples of different alias application to know more how to use different aliases and this dial rule.	
Call Mode	Select differenct signal protocol, SIP or IAX2
Suffix	Set suffix, this is optional config item. It will show no suffix if you don't set it.
Delete Length	Set delete length. This is optional config item. For example: if the delete length is 3, the phone will delete the first 3 digits then send out the rest digits. You can refer to examples of different alias application to know how to set delete length.

Introduction of how to set up dial-peer to implement switch between multi- SIP lines

Number	Destination	Port	Mode	Alias	Suffix	Del Length
9T	0.0.0.1	5060	SIP	no alias	no suffix	0
8T	0.0.0.2	5060	SIP	no alias	no suffix	0

9T mapping: If you have registered a SIP1 server and set dial-peer according to the above table，all calls will be sent via SIP1 server when you press the numeric key "9" in front of dialing destination phone numbers.

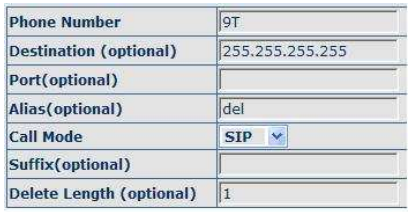
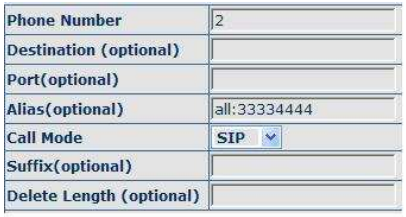
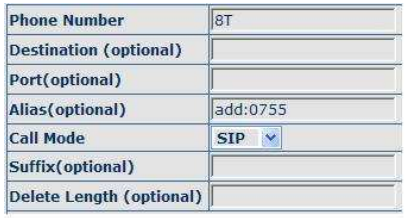
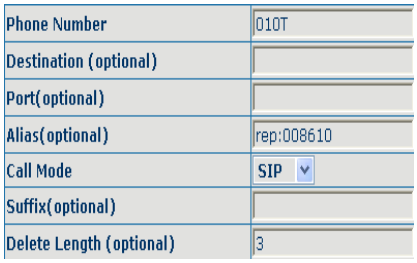
8T mapping: If you have registered a Private SIP2 server and set dial-peer according to the above table，all calls will be sent via SIP2 server when you press the numeric key "8" in front of dialing destination phone numbers.

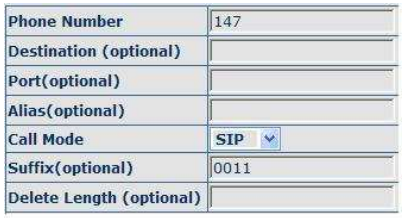
Number	Destination	Port	Mode	Alias	Suffix	Del length
2T	0.0.0.0	4569	IAX2	del	no suffix	1

the rule of 2T means user need to dial the number with prefix 2 if he want to dial via IAX2 server

Examples of different alias application

Set by web	explanation	example

	You need set phone number, Destination, Alias and Delete Length. Phone number is XXXT, Destination is 255.255.255.255 and Alias is del. This means any phone No. that starts with your set phone number will be sent via SIP2 line after the first several digits of your dialed phone number are deleted according to delete length.	If you dial “93333”, the SIP2 server will receive “3333”
	This setting will realize speed dial function, after you dialing the numeric key “2”, the number after all will be sent out.	When you dial “2”, the SIP1 server will receive 33334444
	The phone will automatically send out alias number adding your dialed number, if your dialed number starts with your set phone number.	When you dial “8309“, the SIP1 server will receive “07558309”
	You need set Phone Number, Alias and Delete Length. Phone number is XXXT and Alias is Rep:xxx If your dialed phone number starts with your set phone number, the first digits same as your set phone number will be replaced by the alias number specified and New phone number will be send out.	When you dial “0106228”, the SIP1 server will receive “0086106228”

	If your dialed phone number starts with your set phone number. The phone will send out your dialed phone number adding suffix number.	When you dial “147”, the SIP1 server will receive “1470011”

4.3.4. Phone

4.3.4.1. DSP Config

In this page, you can configure voice codec, input/output volume and so on.

DSP Configuration	
Field name	**explanation**

First Codec	The fist preferential DSP codec: G.711A/u, G.722, G.723, G.729
Second Codec	The second preferential DSP codec: G.711A/u, G.722, G.723, G.729
Third Codec	The third preferential DSP codec: G.711A/u, G.722, G.723, G.729
Forth Codec	The forth preferential DSP codec: G.711A/u, G.722, G.723, G.729
Fifth Codec	The fifth preferential DSP codec: G.711A/u, G.722, G.723, G.729
Input Volume	Specify Input (MIC) Volume grade.
Handfree Volume	Specify Handfree Volume grade
G729 Payload Length	Set G729 Payload Length
Handdown Time	Specify the least reflection time of Handdown, the default is 200ms.
Output Volume	Specify Output (receiver) Volume grade.
Ring Volume	Specify Ring Volume grade
G722 Timestamps	160/20ms or 320/20ms is available
G723 Bit Rate	5.3kb/s or 6.3kb/s is available
Default Ring Type	Set up the ring by default
Signal Standard	Select Signal Standard.
VAD	Select it or not to enable or disable VAD. If enable VAD, G729 Payload length could not be set over 20ms.

4.3.4.2. Call Service

In this web page, you can configure Hotline, Call Transfer, Call Waiting, 3 Ways Call, Black List, white list Limit List and so on.

Call Service	
Field name	**explanation**
Hotline	Specify Hotline number. If you set the number, you can not dial any other numbers.
No Answer Time	Specify No Answer Time
P2P IP Prefix	Set Prefix in peer to peer IP call. For example: what you want to dial is 192.168.1.119, If you define P2P IP Prefix as 192.168.1., you dial only #119 to reach 192.168.1.119. Default is ".". If there is no "." Set, it means to disable dialing IP.
Remote Record No	Set Remote Record number. Via dialing this number, you can hear all voice records in your VoIP server.
Do Not Disturb	Select NO Disturb, the phone will reject any incoming call, the callers

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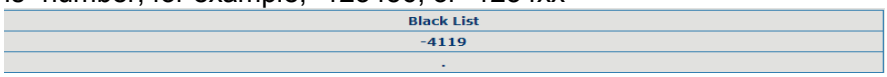
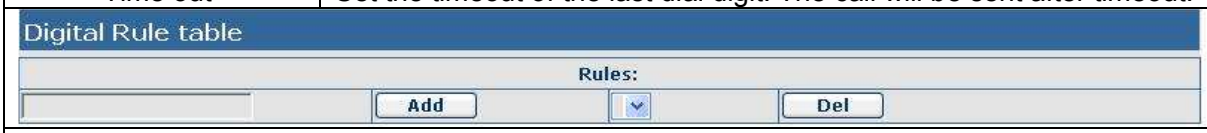
	will be reminded by busy, but any outgoing call from the phone will work well.
Ban Outgoing	If you select Ban Outgoing to enable it, and you can not dial out any number.
Enable Call Transfer	Enable Call Transfer by selecting it.
Enable Call Waiting	Enable Call Waiting by selecting it.
Enable Three Way Call	Enable Three Way Call
Accept Any Call	If select it, the phone will accept the call even if the called number is not belong to the phone.
Auto Answer	If select it, the phone will auto answer when there is an incoming call.
Use Record Server	Select it or not to Enable or disable Use Record Server.
Black List	Set Add/Delete Black list. If user does not want to answer some phone calls, add these phone numbers to the Black List, and these calls will be rejected. x and . are wildcard. x means matching any single digit. for example, 4xxx expresses any number with prefix 4 which length is 4 will be forbidden to dialed out DOT (.) means matching any arbitrary number digit. for example, 6. expresses any number with prefix 6 will be forbidden to dialed out. if user wants to allow a number or a series of number incoming, he may add the number(s) to the list as the white list rule. the configuration rule is -number, for example, -123456, or -1234xx  Means any incoming number is forbidden except for 4119 Note: End with DOT (.) when set up the white list
Limit List	Set Add/Delete Limit List. Please input the prefix of those phone numbers which you forbid the phone to dial out. For example, if you want to forbid those phones of 001 as prefix to be dialed out, you need input 001 in the blank of limit list, and then you can not dial out any phone number whose prefix is 001. x and . are wildcard. x means matching any single digit. for example, 4xxx expresses any number with prefix 4 which length is 4 will be forbidden to dialed out . means matching any arbitrary number digit. for example, 6. expresses any number with prefix 6 will be forbidden to dialed out.
Notice: Black List and Limit List can record at most10 items respectively.	

4.3.4.3. Digital Map Configuration

This phone supports 4 dial modes:
1). End with "#": dial your desired number, and then press #.
2). Fixed Length: the phone will intersect the number according to your specified length.
3). Time Out: After you stop dialing and waiting time out, system will send the number collected.
4). User defined: you can customize digital map rules to make dialing more flexible. It is realized by defining the prefix of phone number and number length of dialing.

In order to keep some users' secondary dialing manner when dialing the external line with pbx, phone can be added a special rule to realize it. so user can dial a number as external line prefix and get the secondary dial tone to keep dial the external number. after finishing dialing, phone will send the prefix and external number totaly to ther server.
for example, there is a rule 9,xxxxxxxx in the digital map table. after dialing 9, phone will send the secondary dial tone, user may keep going dialing. after finished, phone will call the number which starts with 9, actually the number sent out is 9-digit with 9.

Digital Map Configuration	
Field name	**explanation**
End with "#"	Set Enable/Disable the phone ended with “#” dial.
Fixed Length	Specify the Fixed Length of phone ending with.
Time out	Set the timeout of the last dial digit. The call will be sent after timeout.



Below is user-defined digital map rule:
[] Specifies a range that will match digit. May be a range, a list of ranges separated by commas, or a list of digits.
x Match any single digit that is dialed.
. Match any arbitrary number of digits including none.
Tn Indicates an additional time out period before digits are sent of n seconds in length. n is mandatory and can have a value of 0 to 9 seconds. Tn must be the last 2 characters of a dial plan. If Tn is not specified it is assumed to be T0 by default on all dial plans.

RULE
"[1-8]xxx"
"9xxxxxxx"
"911"
"99T4"
"9911x.T4"

[
1-8]xxx: Cause extensions 1000-8999 to be dialed immediately
9xxxxxxx: Cause 8 digit numbers started with 9 to be dialed immediately
911: Cause 911 to be dialed immediately after it is entered.
99T4: Cause 99 to be dialed after 4 seconds.
9911x.T4:Cause any number started with 9911 to be dialed 4 seconds after dialing ceases.

Notice: End with “#”, Fixed Length, Time out and Digital Map Table can be used simultaneously, System will stop dialing and send number according to your set rules.

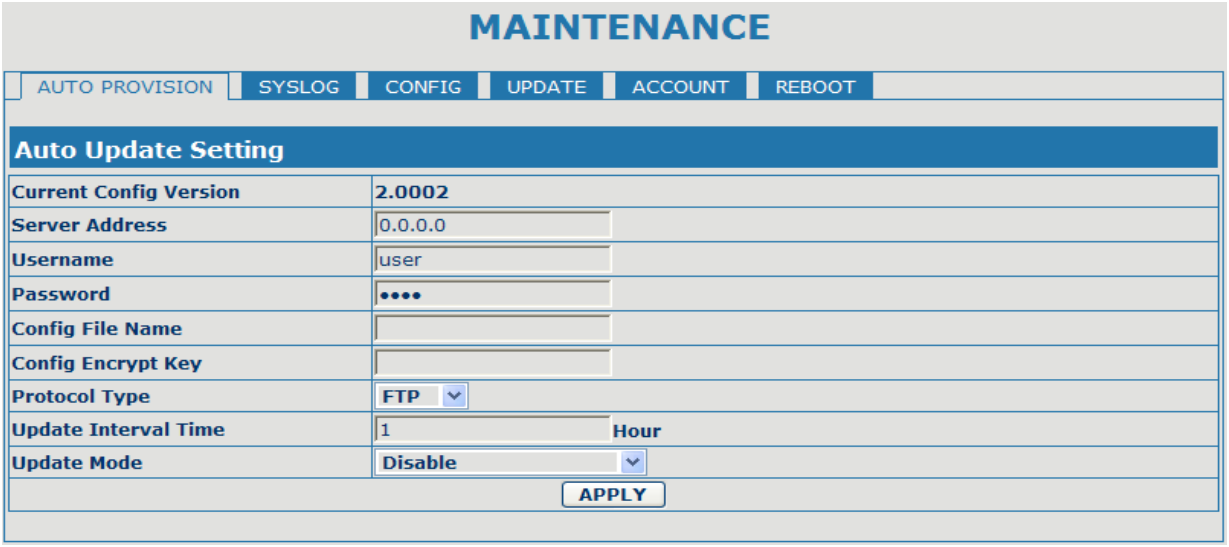
4.3.4.4. Phone Book

You can input the name, phone number and select ring type for each name here.

Phone Book	
Field name	**explanation**
Index / Name / Number / Type 1 / ad / 23 / Default 1	Shows the detail of current phonebook.
Name	Shows the name corresponding to the phone number
Number	Shows the phone number
Ring Type	Shows the ring type of the incoming call.
Click “Modify” to change the selected information and click the “Delete” to delete the selected record.	
Notice: the maximum capability of the phonebook is 500 items	

4.3.5. Maintenance

4.3.5.1. Auto Provision



Auto Provision	
Field name	**explanation**
Current Config Version	Show the current config file's version.
Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be IP address or Domain name with subdirectory.
Username	Set FTP server Username. System will use anonymous if username keep blank.
Password	Set FTP server Password.
Config File Name	Set configuration file's name which need to update. System will use MAC as config file name if config file name keep blank. For example, 000102030405.。
Config Encrypt Key	Input the Encrypt Key, if the configuration file is encrypted.
Protocol Type	Select the Protocol type FTP、TFTP or HTTP.
Update Interval Time	Set update interval time, unit is hour.
Update Mode	Different update modes: 1. Disable: means no update 2. Update after reboot: means update after reboot. 3. Update at time interval: means periodic update.

4.3.5.2. Syslog Config

Syslog is a protocol which is used to record the log messages with client/server mechanism. Syslog server receives the messages from clients, and classifies them based on priority and type. Then these messages will be written into log by some rules which administrator can configure. This is a better way for log management.

8 levels in debug information:

Level 0---emergency: This is highest default debug info level. You system can not work.

Level 1---alert: Your system has deadly problem.

Level 2---critical: Your system has serious problem.

Level 3---error: The error will affect your system working.

Level 4---warning: There are some potential dangers. But your system can work.

Level 5---notice: Your system works well in special condition, but you need to check its working environment and parameter.

Level 6---info: the daily debugging info.

Level 7---debug: the lowest debug info. Professional debugging info from R&D person.

At present, the lowest level of debug information send to Syslog is info, debug level only can be displayed on telnet.

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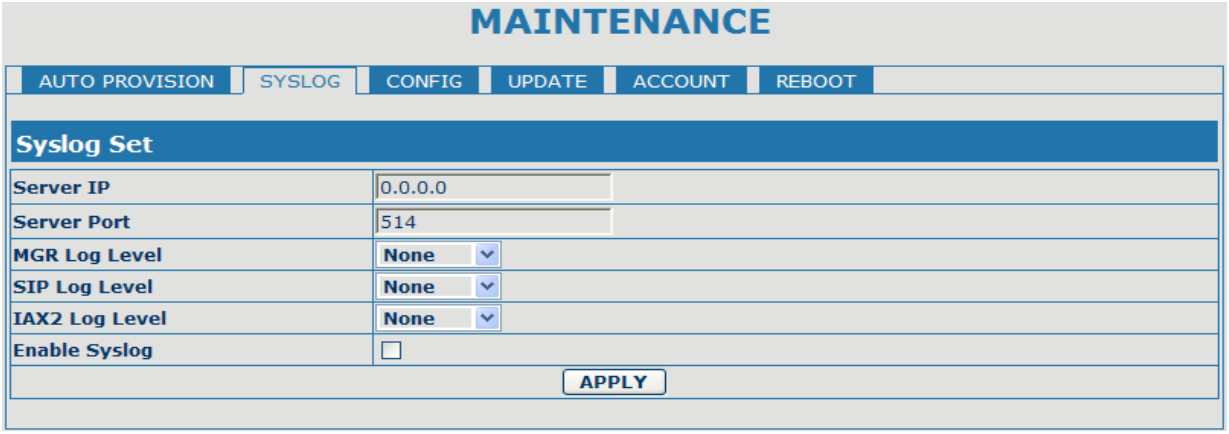
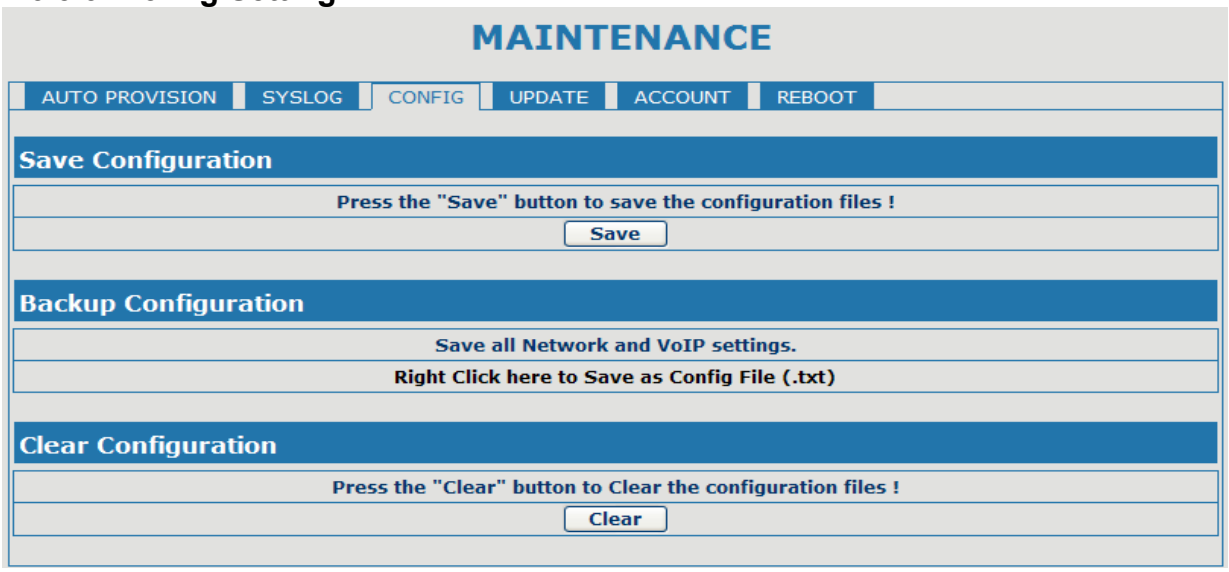


Syslog Configuration	
Field name	**explanation**
Server IP	Set Syslog server IP address.
Server Port	Set Syslog server port.
MGR Log Level	Set the level of MGR log.
SIP Log Level	Set the level of SIP log.
IAX2 Log Level	Set the level of IAX2 log.
Enable Syslog	Select it or not to enable or disable syslog.

4.3.5.3. Config Setting



Config Setting	
Field name	**explanation**
Save Config	you can save all changes of configurations. Click the Save button, all changes of configuration will be saved, and be effective immediately. .
Backup Config	Right clicks on “Right click here…” and select “Save Target As….” then you will save the config file in .txt format
Clear Config	user can restore factory default configuration and reboot the phone. If you login as Admin, the phone will reset all configurations and restore factory default; if you login as Guest, the phone will reset all configurations except for VoIP accounts (SIP1-2 and IAX2) and version number.

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4.3.5.4. Update

You can update your configuration with your config file in this web page.

Update	
Field name	**explanation**
Web Update	Click the browse button, find out the config file saved before or provided by manufacturer, download it to the phone directly, press “Update” to save. You can also update downloaded update file, logo picture, ring, mmiset file by web.
Server	Set the FTP/TFTP server address for download/upload. The address can be IP address or Domain name with subdirectory.
Username	Set the FTP server Username for download/upload.
Password	Set the FTP server password for download/upload.
File name	Set the name of update file or config file. The default name is the MAC of the phone, such as 000102030405.
Notice: You can modify the exported config file. And you can also download config file which includes several modules that need to be imported. For example, you can download a config file just keep with SIP module. After reboot, other modules of system still use previous setting and are not lost.	
Type	Action type that system want to execute： 1. Application update: download system update file 2. Config file export: Upload the config file to FTP/TFTP server, name and save it. 3. Config fie import: Download the config file to phone from FTP/TFTP server. The configuration will be effective after the phone is reset.
Protocol	Select FTP/TFTP server

4.3.5.5. Account Config

You can add or delete user account, and change the authority of each user account in this web page

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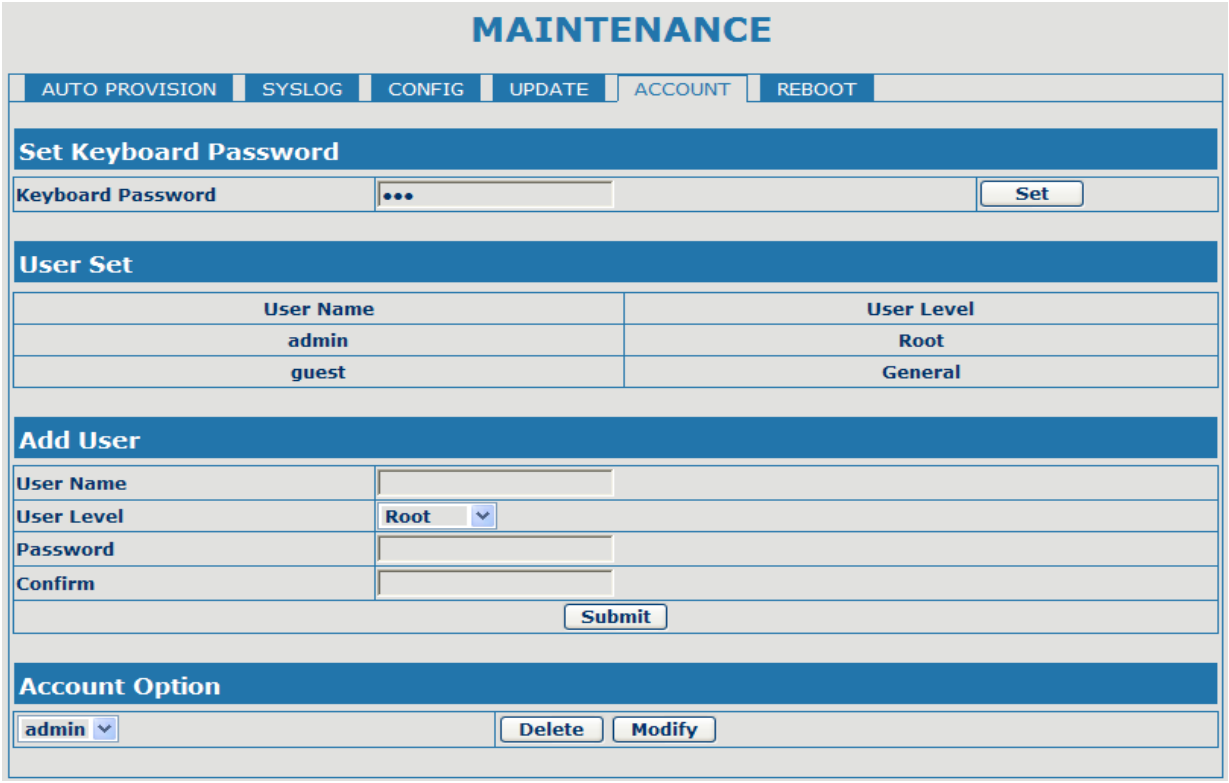
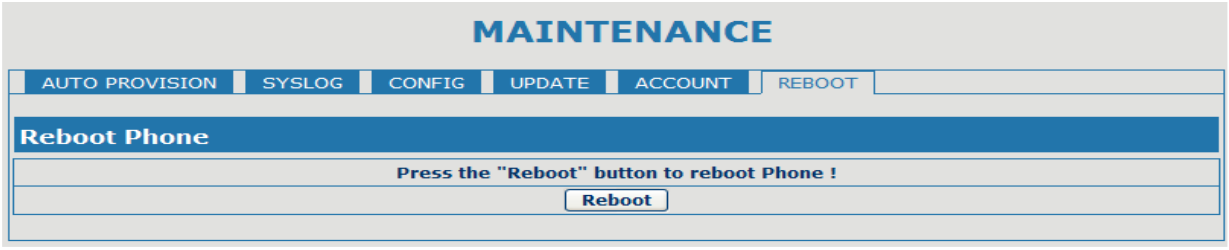


Account Configuration	
Field name	**explanation**
Keyboard Password	Set the password for entering the setting menu of the phone by the phone ‘s key board. The password is digit.
User Name / User Level admin / Root guest / General	This table shows the current user existed.
User Name	Set account user name.
User Level	Set user level, Root user has the right to modify configuration, General can only read.
Password	Set the password.
Confirm	Confirm the password.
Select the account and click the **Modify** to modify the selected account, and click the **Delete** to delete the selected account. General user only can add the user whose level is General.	

4.3.5.6. Reboot



If you modified some configurations which need the phone’s reboot to be effective, you need click the Reboot, then the phone will reboot immediately.
Notice: Before reboot, you need confirm that you have saved all configurations..

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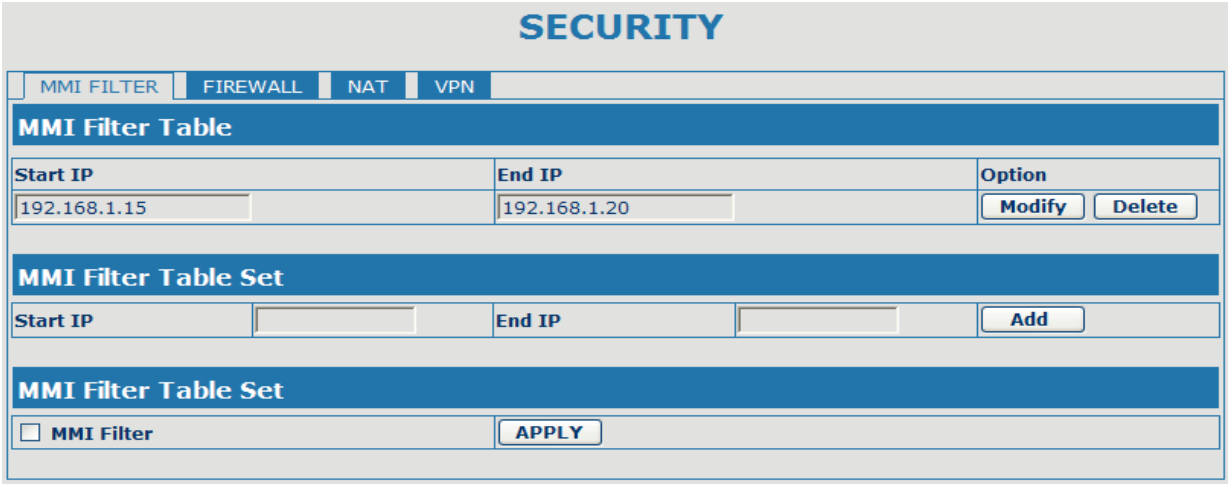
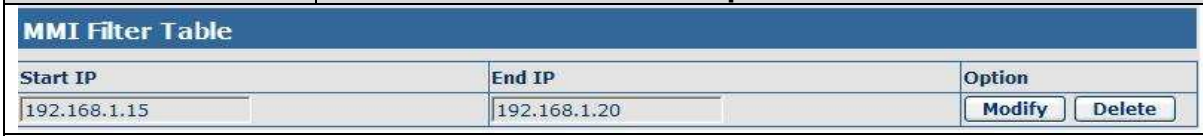
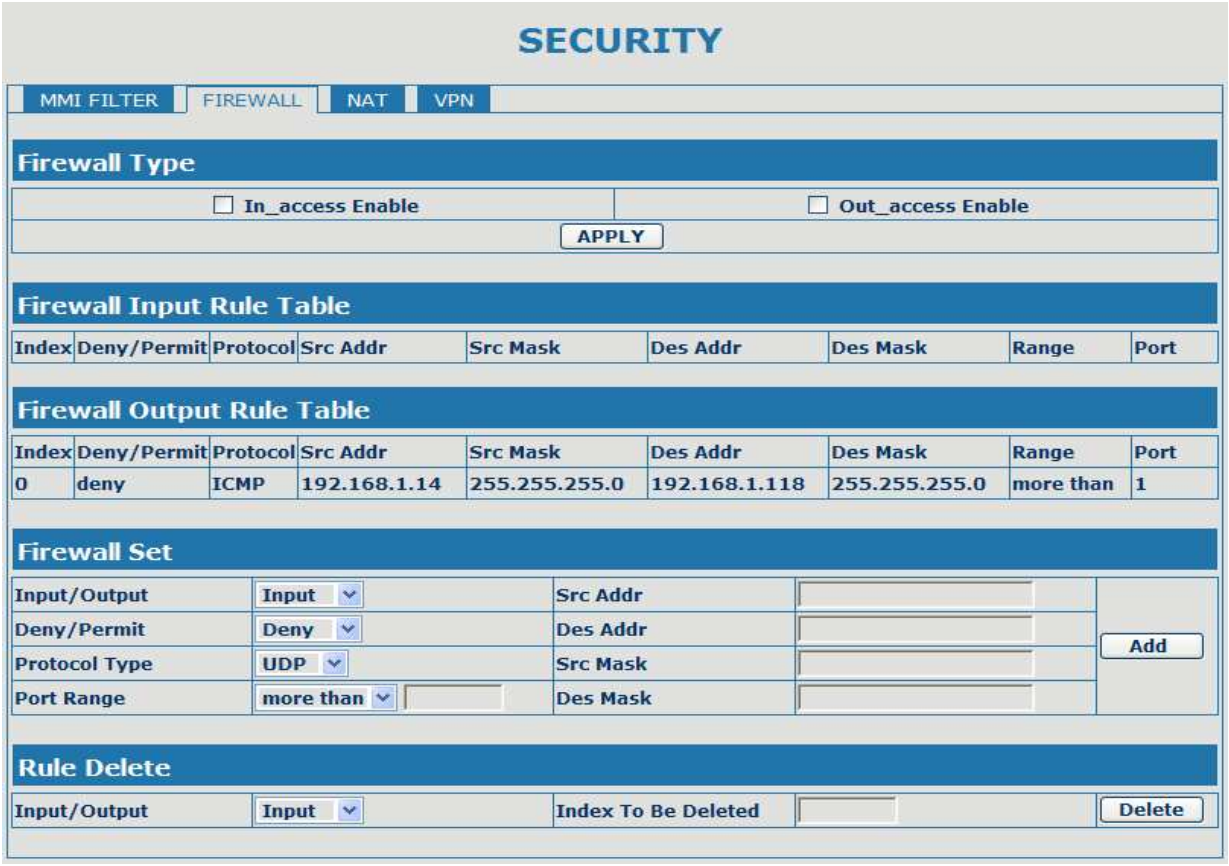
4.3.6. Security

4.3.6.1. MMI Filter



MMI Filter	
User could make some device own IP, which is pre-specified, access to the MMI of the phone to config and manage the phone.	
Field name	**explanation**
	
MMI Fileter IP Table list:	
Add or delete the IP address segments that access to the phone. Set initial IP address in the Start IP column, Set end IP address in the End IP column, and click Add to add this IP segment. You can also click Delete to delete the selected IP segment.	
MMI Filter	Select it or not to enable or disable MMI Filter. Click **Apply** to make it effective.
Notice: Do not set your visiting IP outside the MMI filter range, otherwise, you can not logon through the web.	

4.3.6.2. Firewall



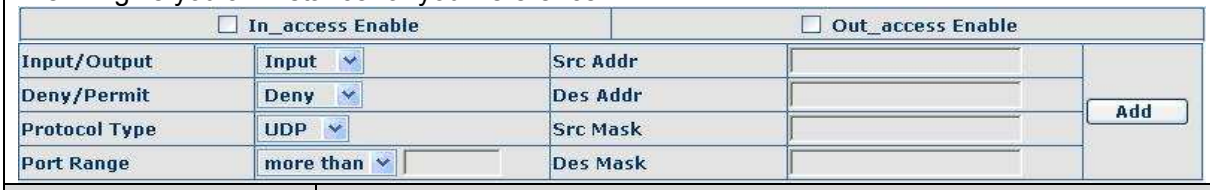
Firewall Configuration

In this web interface, you can set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices from accessing the Internet (output rule).

Firewall supports two types of rules: input_access rule and output_access rule. Each type supports at most 10 items.

Through this web page, you could set up and enable/disable firewall with input/output rules. System could prevent unauthorized access, or access other networks set in rules for security. Firewall, is also called access list, is a simple implementation of a Cisco-like access list (firewall). It supports two access lists: one for filtering input packets, and the other for filtering output packets. Each kind of list could be added 10 items.

We will give you an instance for your reference.



Field name	explanation
In_access enable	Select it to Enable in_ access rule
out_access enable	Select it to Enable out_ access rule
Input/Output	Specify current adding rule by selecting input rule or output rule.
Deny/Permit	Specify current adding rule by selecting Deny rule or Permit rule.
Protocol Type	Filter protocol type. You can select TCP, UDP, ICMP, or IP.
Port Range	Set the filter Port range
Src Addr	Set source address. It can be single IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.0
Des Addr	Set the destination address. It can be IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.*

Src Mask	Set the source address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.
Des Mask	Set the destination address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.

Click the **Add** button if you want to add a new output rule.

Firewall Output Rule Table

Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port
0	deny	ICMP	192.168.1.14	255.255.255.0	192.168.1.118	255.255.255.0	more than	1

Then enable out_access, and click the Apply button.
So when devices execute to ping 192.168.1.118, system will deny the request to send icmp request to 192.168.1.118 for the out_access rule. But if devices ping other devices which network ID is 192.168.1.0, it will be normal.

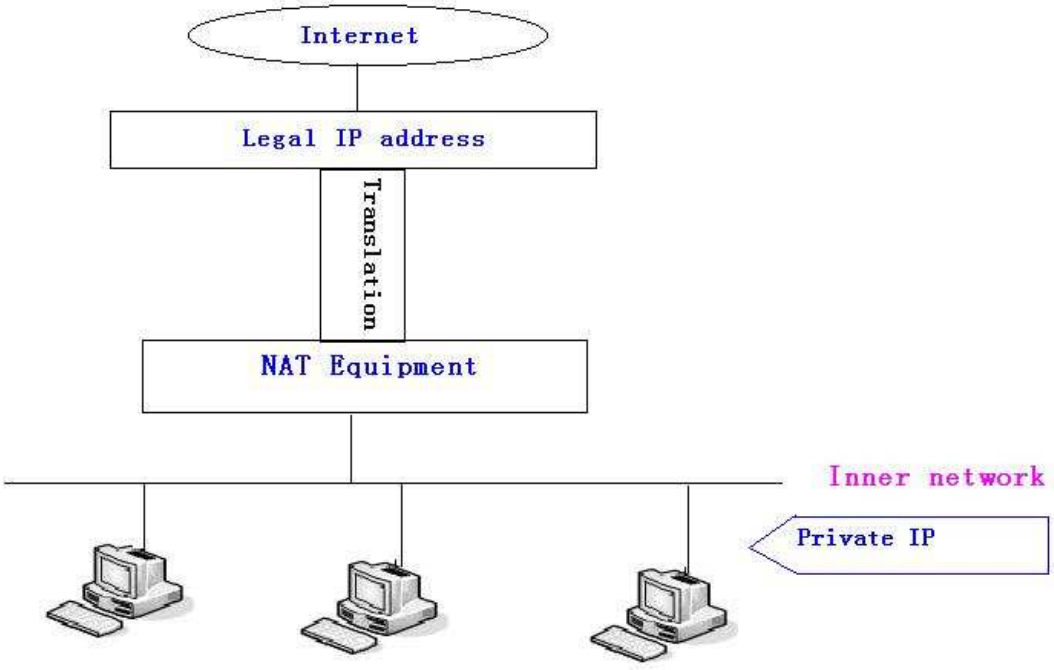
Rule Delete

Input/Output	Input	Index To Be Deleted		Delete

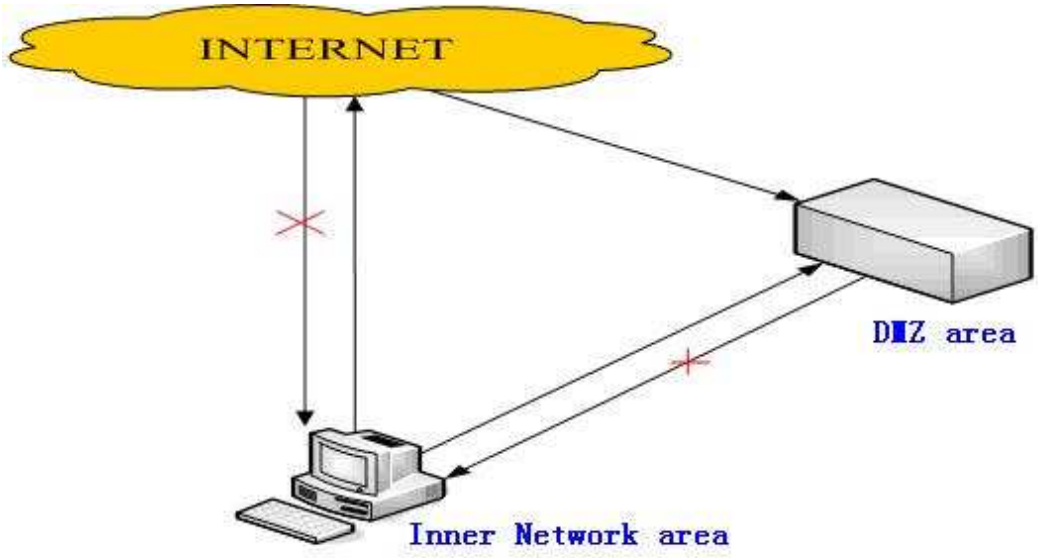
Click the **Delete** button to delete the selected rule.

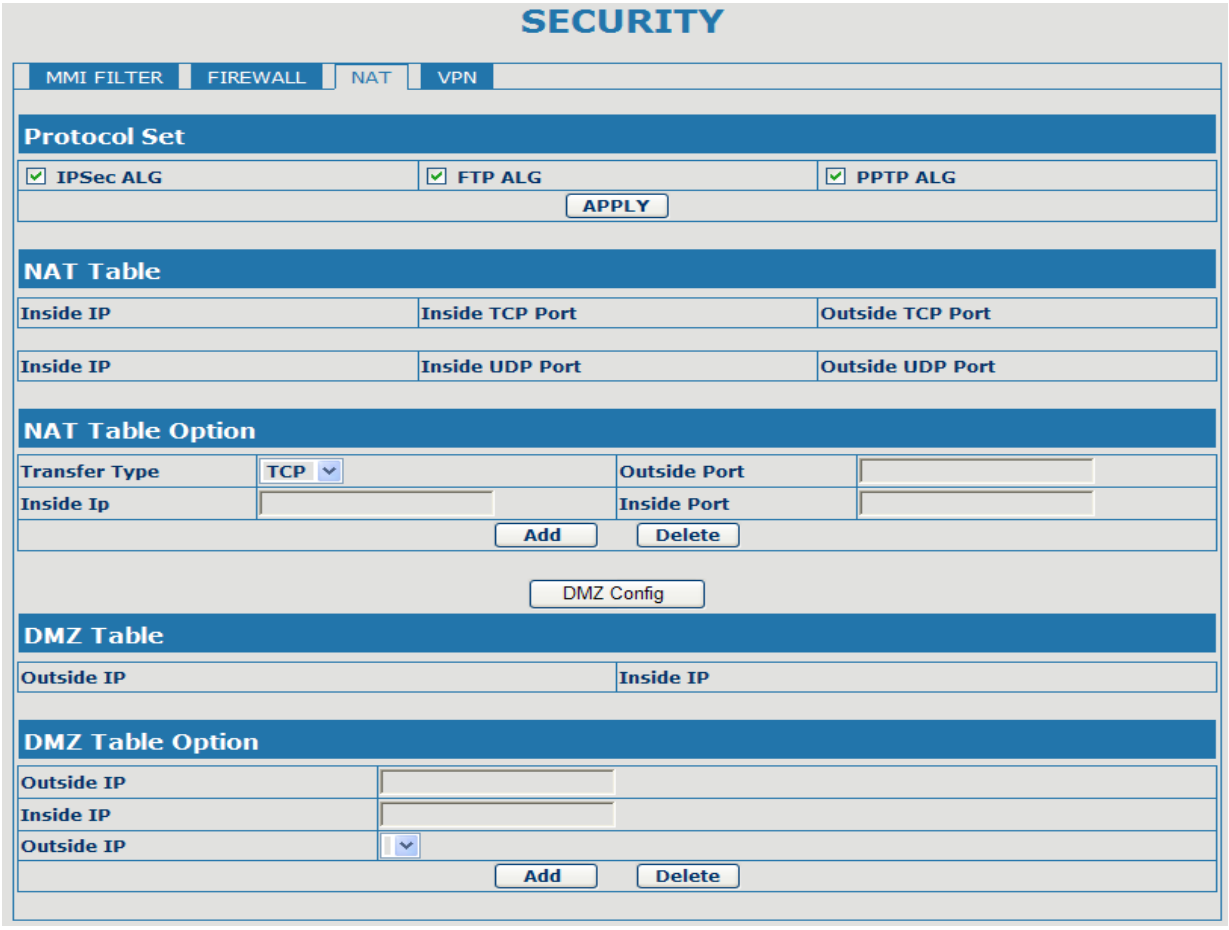
4.3.6.3. NAT Config

NAT is abbreviated from Net Address Translation; it's a protocol responsible for IP address translation. In other word, it is responsible for transforming IP and port of private network to public, also is the IP address mapping which we usually say.



DMZ config：
In order to make some intranet equipments support better service for extranet, and make internal network security more effectively, these equipments open to extranet need be separated from the other equipments not open to extranet by the corresponding isolation method according to different demands. We can provide the different security level protection in terms of the different resources by building a DMZ region which can provide the network level protection for the equipments environment, reduce the risk which is caused by providing service to distrust customer, and is the best position to put public information
The following chart describes the network access control of DMZ





NAT Configuration	
Field name	**explanation**
IPSec ALG	It is an encryption technology. Select it to enable IPSec ALG, the default is enable
FTP ALG	FTP is a service of connection layer which can transform intranet IP into extranet IP when intranet IP is sending out packet. Select it to enable FTP ALG, the default is enable
PPTP ALG	Select it enable PPTP ALG, the default is enable

Inside IP	Inside TCP Port	Outside TCP Port

Shows the NAT TCP mapping table

Inside IP	Inside UDP Port	Outside UDP Port

Shows the NAT UDP mapping table

NAT Table Option

Transfer Type	TCP	Outside Port	
Inside Ip		Inside Port	

Add Delete

Transfer Type	Select the NAT mapping protocol style, TCP or UDP
Inside IP	Set the IP address of device which is connected to LAN interface to do NAT mapping.
Inside Port	Set the LAN port of the NAT mapping
Outside Port	Set the WAN port of the NAT mapping

Notice: After finish setting, click the Add button to add new mapping table; click the Delete button to delete the selected mapping table.

DMZ Table

Outside IP	Inside IP
192.168.1.119	192.168.10.23

Shows the outside WAN port IP address and the inside LAN port IP address.

Outside IP	
Inside IP	
Outside IP	192.168.1.119

Add Delete

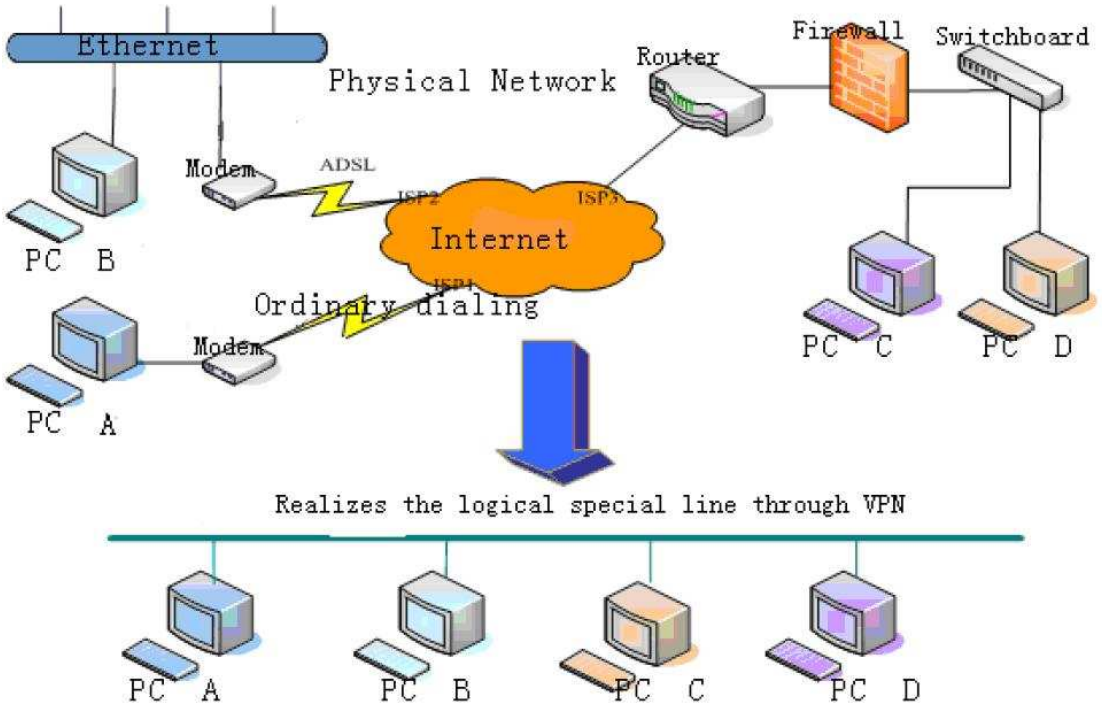
Outside IP	Set the outside Wan port IP address of DMZ.
Inside IP	Set the inside LAN port IP address of DMZ

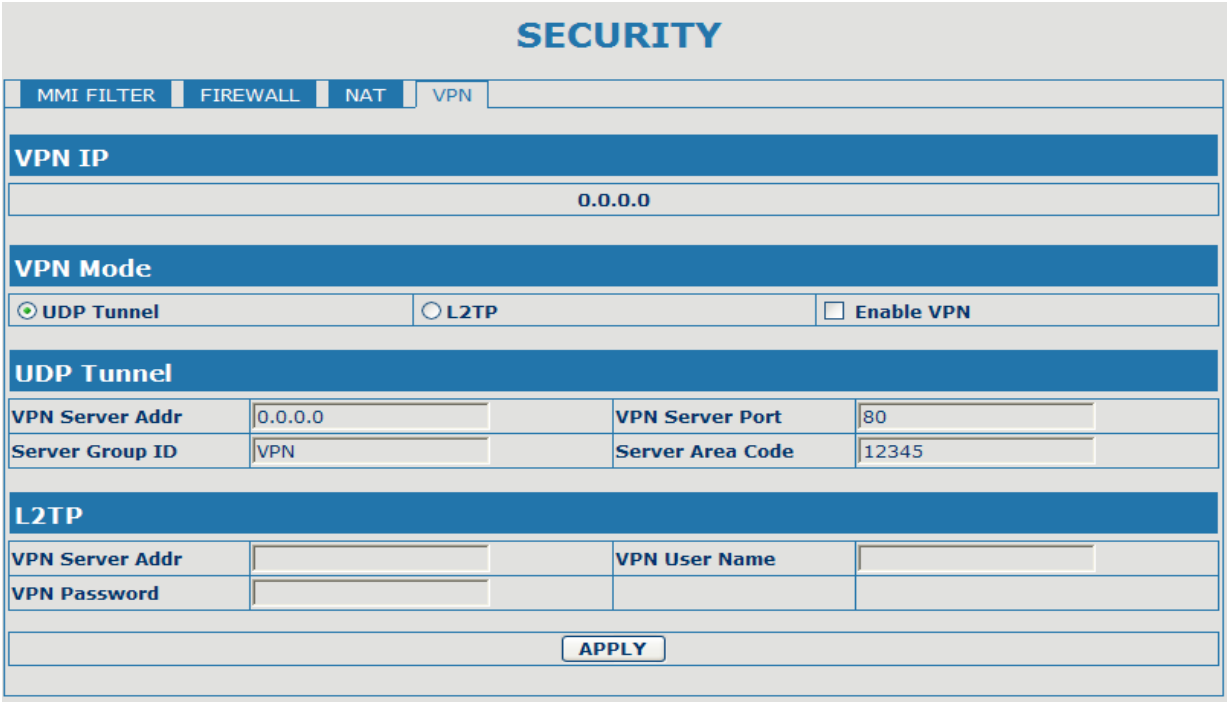
Click the **Add** button to add new table; click the **Delete** button to delete the selected mapping table.

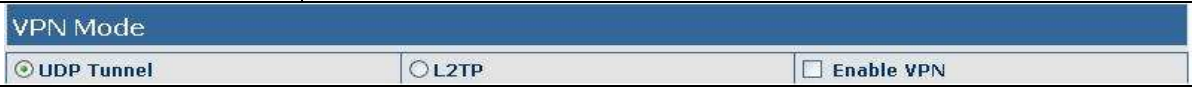
Notice: 10M/100M adaptive means the network card, and other equipment physical consultations speed, testing speed under bridge mode near to 100M, in order to ensure the quality of voice and communications real-time performance, we made some sacrifices of NAT under the transmission performance. Transmit with full capability only when system is idle, so can not guarantee that the transmission speed reach to 100M.

4.3.6.4. VPN Config

This web page provides us a safe connect mode by which we can make remote access to enterprise inner network from public network. That is to say, you can set it to connect public networks in different areas into inner network via a special tunnel.





VPN Configuration	
Field name	**explanation**
VPN IP	Shows the current VPN IP address
	
Select UDP Tunnel (VPN Tunnel) or VPN L2TP. You can choose only one for current state. After you select it, you'd better save configuration and reboot your phone.	
Enable VPN	Select it or not to enable or disable VPN;

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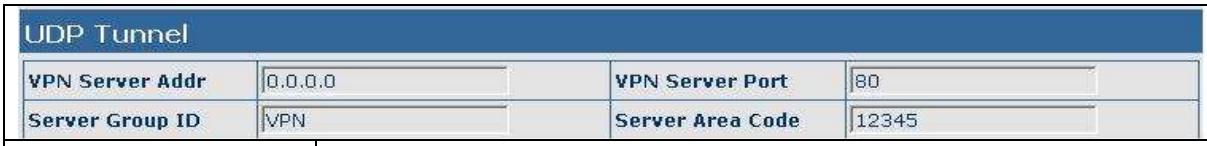
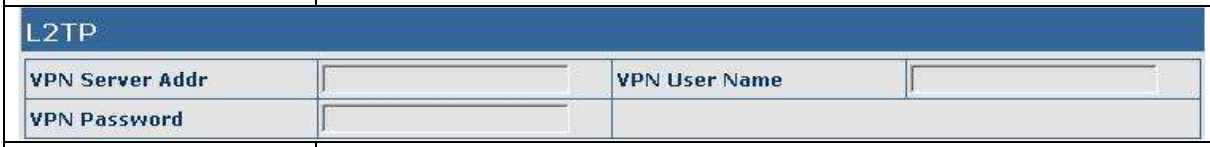
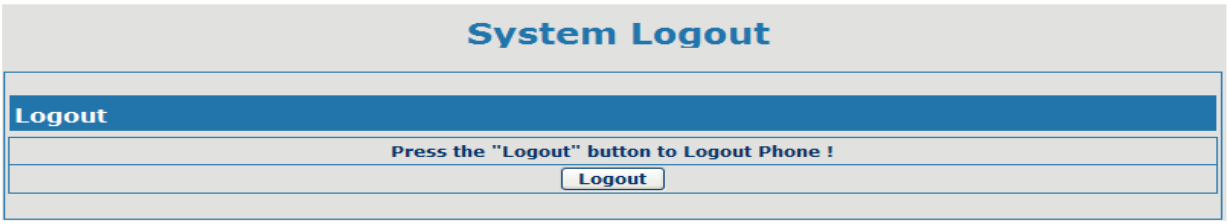


VPN Server Addr	Set VPN Server IP Address
VPN Server Port	Set VPN Server Port



VPN Server Addr	Set VPN L2TP Server IP address
VPN User Name	Set User Name access to VPN L2TP Server
VPN Password	Set Password access to VPN L2TP Server

4.3.7. Logout



Click **Logout**，and you will exit web page. If you want to enter it next time, you need input user name and password again.

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4.4. Settings via phone's keyboard.

4.4.1. How to set via the phone's keyboard.

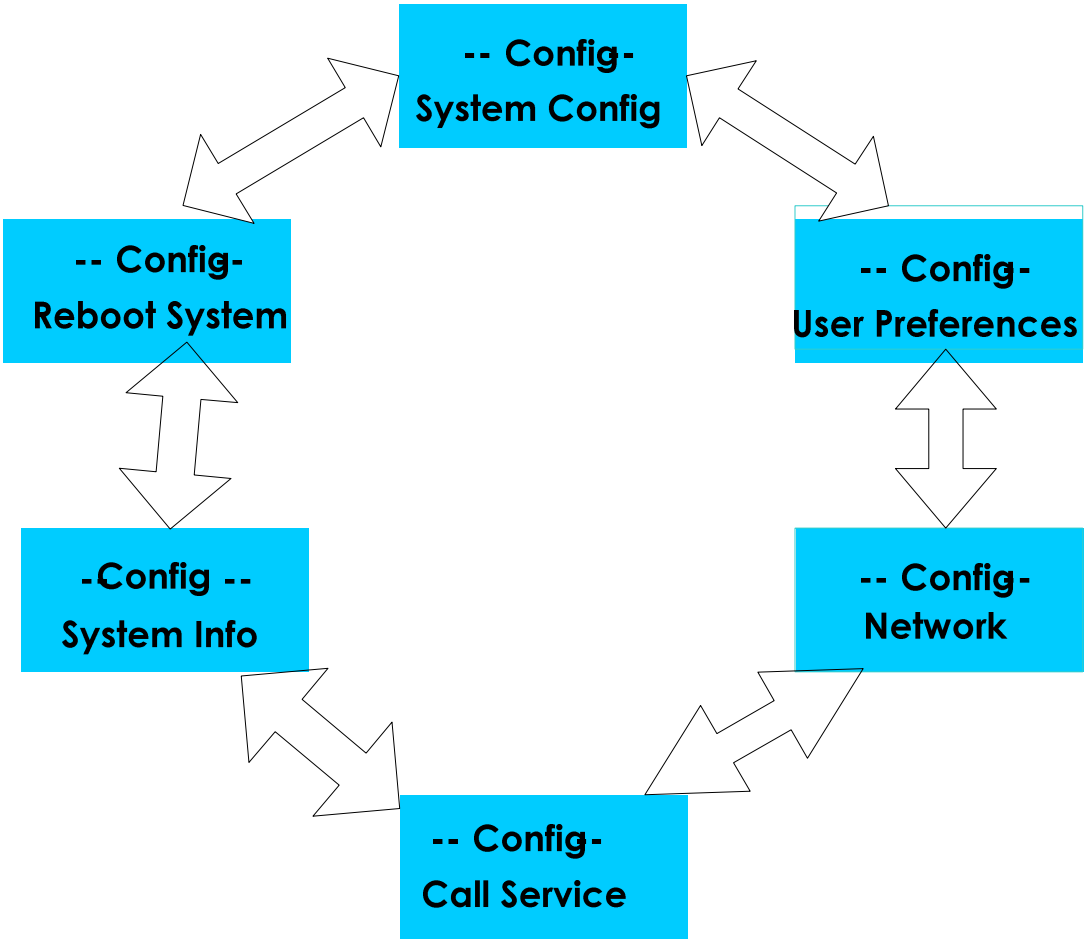
User can do browse, modify or cancel via screen menu by using

, and Soft1/Soft2/Soft3.

- Use and Soft1 to select the sub-menu.
- Use to adjust screen brightness and contrast, ring volume and voice volume.
- Use Soft2 and Soft3 to enter/modify or exit/cancel.

4.4.2. Phone menu

Phone main menu:



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5. Appendix

5.1. Specification

5.1.1. Device specification

Item		this VoIP Phone
Adapter(Input/Output)		Input：100-240VAC 50-60Hz Output：DC 5V/1A
Port	WAN	10/100Base- T RJ-45 for LAN
	LAN	10/100Base- T RJ-45 for PC
Power Consumption		Idle：1.5W/Active：1.8W
LCD size		74 x 28mm dot-matrix LCD
Operation Temperature		0～40℃
Relative Humidity		10～65%
Main Chipset		BCM 1190
SDRAM		16Mbits
Flash		4Mbits
Size（W x H x D）		11.6×8×3 in.(295×205×75mm)
Weight		2.07lb.(0.94kg)

5.1.2. Voice Features

- Support 2 lines SIP and IAX2, SIP 2.0 (RFC3261)
- Codec：G.711A/u，G.723 high/low, G.729, G.722
- Echo cancellation： Support G.168 and hand-free can support 96ms
- Support VAD，CNG
- NAT transverse: support STUN
- Supports full duplex.
- SIP support SIP domain，SIP authentication（none，basic， MD5），DNS name of server, peer to peer
- SIP support 2 SIP lines. Can connect to SIP1 and SIP2 server at the same time
- DTMF：SIP info，DTMF Relay，RFC2833
- SIP application: support Call forward/ transfer/ holding/ waiting / 3 way talking/ paging and intercom/pickup/join call/redial/unredial/push to talk/click to dial/call park
- Call control features: Flexible dial map, support hotline, empty calling no. reject server, black list for reject, authenticated call, no disturb, caller ID and so on.
- Support phonebook 500 records, incoming calls / outgoing calls / missing calls. Each supports 100 records
- support conference call and record on server
- Could dial use private server automatically when public server unregistered while private server is resgistered successfully
- Phonebook supports VCard standard
- Support 12/24 time format.
- 12/24 hours time display
- Support daylight saving time
- Support path, gruu
- Support SIP Privacy.
- Support IAX2
- Support 9 kind of ring type

5.1.3. Network Features

- WAN/LAN: support Bridge and Router mode.

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- Support basic NAT and NAPT
- Support PPPoE for xDSL
- support VLAN
- Support NAT penetration, and Stun penetration
- Support DMZ
- Support VPN(L2TP. UDP)
- Support DHCP get IP on WAN port
- Support DHCP distribute IP on LAN port
- Qos supports Diffserv.
- support network tools: contain ping，trace route，telnet client

5.1.4. Maintenance and Management

- The phone supports post mode, can update firmware by post mode.
- Supports different levels of administration.
- Support Boot Monitor
- Can upgrade firmware through boot monitor
- access with different authority
- support auto provisioning(upgrade firmware or configuration file)
- Can config through Web, Keypad, Telnet
- Can upgrade firmware and configuration file through HTTP, FTP, TFTP
- Support syslog

5.2. Digit-character map table

Button	Character	Button	Character
1	1 @	7 PQRS	7 P Q R S p q r s
2 ABC	2 A B C a b c	8 TUV	8 T U V t u v
3 DEF	3 D E F d e f	9 WXYZ	9 W X Y Z w x y z
4 GHI	4 G H I g h i	*\/■	.
5 JKL	5 J K L j k l	0 OPER	0
6 MNO	6 M N O m n o	#/=	#